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Symetrix

602

Stereo Digital Processor

Owner's Manual

Manual: Rev 1.1, 11/15/94
Software: Rev 2.03
Part number: 530602
Subject to change at our whim, and without notice.

Batteries not included. Ground isn't ground!
Available at finer studios everywhere.

Production Information

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Ain't technology grand?

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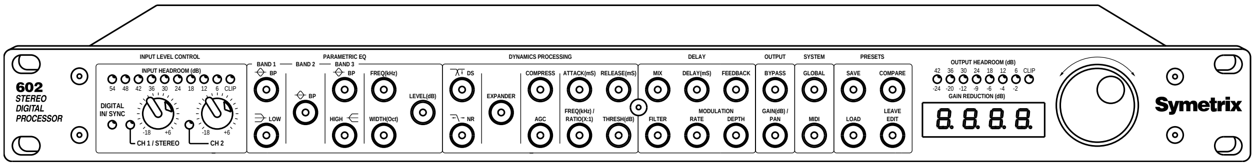


1. Introduction

The Symetrix 602 Stereo Digital Processor is a dual-channel digital signal processor intended for use in a variety of recording, broadcast, live sound and post production applications. Acting as a "bridge" between the analog and digital domains, the 602 accepts stereo line level analog signals, converts them to 18-bit digital (44.1 kHz or 48 kHz sample rates), performs 24-bit digital signal processing, and sends them on their way via the digital and analog outputs. The 602 uses two Motorola DSP-56001 digital signal processors (DSP) for an overall processing rate of 40 million instructions per second (40 MIPS).

The 602 has inputs and outputs accommodating all common analog and digital formats. The following table lists all of the inputs and outputs.

Input	Mode	Output	Mode
Line (x2)	A	Line (x2)	A
AES/EBU	D	AES/EBU	D
S/PDIF	D	S/PDIF	D

The stereo line inputs may be used in various combinations. The input and output modes are separate; you can use almost any combination of the analog and digital outputs simultaneously (for example, the AES/EBU and S/PDIF digital *outputs* cannot be used simultaneously).

While the 602 works great for voice (singing or monologue/dialogue) enhancement, its powerful digital engine works wonders on any signal. Processing includes fully parametric EQ, shelving EQ, notch filtering, dynamic noise filtering, de-essing, delay (first reflection), stereo synthesis, gating, expansion, compression, and AGC (automatic gain control). Get the picture?

One aspect of many digital processors is the difficulty of use. The 602 was designed to be easy to use, yet powerful. There are no menus to scroll through. Each parameter is visible via the front panel push-switches. Pressing a switch transfers the display to that parameter's current value. The parameter wheel allows you to change the value. Finally, the 602 allows you to compare your stored setting with the current (edited) setting, without committing the edited settings to memory.

Of course, all this processing power can be remotely controlled via MIDI. The 602's MIDI implementation includes simple program change as well as parameter editing.

All analog inputs and outputs are available via XLR connectors. The AES/EBU digital inputs and outputs use XLR connectors and the S/PDIF digital inputs and outputs use RCA connectors. The MIDI input and output connections use standard 5-pin female DIN connectors.

The 602's unique set of digital tools can make voices, instruments, or sound effects jump out of any mix.; Its combination of factory presets and non-volatile user program space guarantee predictable and repeatable effects from session to session, performance to performance.

We recommend that you read this manual from cover to cover. Somewhere between the confines of the two covers you should find the answers to most (98%) of your questions, both technical as well as musical.

If you're in a hurry (like most of us), or if you really don't believe that someone could write a decent owners manual that you can read and understand, then do us both a favor and read the remainder of this section and Chapter 6, "Fast First Time Setup." Chapter 6 will help you get

connected, tell you what the knobs do, and send you on your way. For MIDI information, go directly to Appendix C, which describes some of the things that you can do with the 602 using MIDI.

1.1 Manual Sections

This manual contains the following sections:

Chapter 1. *Introduction* introduces the 602 and this manual. Describes important safety information

Chapter 2. *Basics* lets you know what the 602 does, and how it does it and some basic usage information..

Chapter 3. *Technical Tutorial* a basic and not-so-basic discussion of signal levels, input and output impedances, and connection polarity.

Chapter 4. *Front Panel Overview* gives a brief look at the controls and switches located on the front panel of the 602.

Chapter 5. *Rear Panel Overview* gives a brief look at the rear panel of the 602.

Chapter 6. *Fast, First Time Setup* is a section written especially for people who just can't wait to get their hands on the knobs.

Chapter 7. *Using the* 602 describes the use of the 602 in detail.

Chapter 8. *Applications* describes some of the myriad uses for the 602.

Chapter 9. *Troubleshooting* tells what to do if the 602 doesn't work.

Chapter 10. *Limited Warranty* describes the 602's warranty.

Chapter 11. *Repair Information* tells how to get your 602 repaired.

Chapter 12. *Specifications* lists the technical specifications of the 602's performance.

Appendix A. *Appendix A* describes how to use the realtime MIDI features built into the 602.

Appendix B. *Appendix B* tells how to use the Lexicon MRC with the 602.

Appendix C. *Appendix C* describes how to communicate with the 602 via MIDI. This appendix also contains a description of the 602's Midi implementation.

Appendix D. *Appendix D* contains a glossary and a useful bibliography.

Appendix E. *Appendix E* contains the Architects and Engineer's specifications.

Appendix F. *Appendix F* contains disassembly instructions.

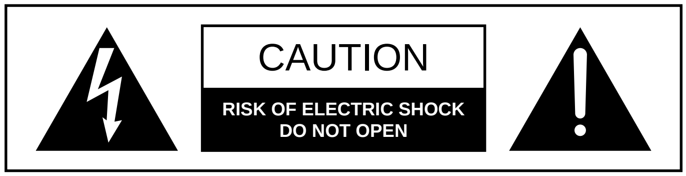
Appendix G. *Appendix G* contains a listing of the preset programs and other miscellany.

1.2 Operator Safety Summary

The information in this summary is intended for persons who operate the equipment as well as repair personnel. Specific warnings and cautions are found throughout this manual wherever they may apply; they do not appear in this summary.

The notational conventions used in this manual and on the equipment itself are described in the following paragraphs.

1.2.1 Equipment Markings



ATTENTION: RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR

No user serviceable parts inside. Refer servicing to qualified service personnel.
Il ne se trouve a l’interieur aucune piece pourvant entre reparée l’usager.
S’adresser a un reparateur compétent.

The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user of the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.	The exclamation point within an equilateral triangle is intended to alert the user of the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance (i.e. this manual).

Caution

To prevent electric shock, do not use the polarized plug supplied with this appliance with any extension cord, receptacle, or other outlet unless the blades can be fully inserted to prevent blade exposure.

1.2.2 Terms

Several notational conventions are used in this manual. Some paragraphs may use Note, Caution, or Warning as a heading. These headings have the following meaning:

Convention	Description
Caution	*Identifies information that, if not heeded, may cause damage to the* 602 *or other equipment in your system.*
Note	Identifies information that needs extra emphasis. A Note generally supplies extra information to help you use the 602 better.
Warning	**Identifies information that, if ignored, may be hazardous to your health or that of others.**

In addition, certain typefaces and capitalization are used to identify certain words. These situations are:

Convention	Meaning
CAPITALS	Controls, switches or other markings on the chassis.
Boldface	Strong emphasis.
Helvetica-Narrow	Information appearing on the LED display

Finally, two symbols are used as visual hints. They are:

Symbol	Meaning
	Helping hand. A hint to make your life a bit easier.
	The Bomb. A visual way of saying, “Caution!”

1.3 Other Safety Information

Power Source	This product is intended to operate from a power source that does not apply more than 250V rms between the power supply conductors or between either power supply conductor and ground. A protective ground connection, by way of the grounding conductor in the power cord, is essential for safe operation
Grounding	The chassis of this product is grounded through the grounding conductor of the power cord. To avoid electric shock, plug the power cord into a properly wired receptacle before making any connections to the product. A protective ground connection, by way of the grounding conductor in the power cord, is essential for safe operation.
Danger from Loss of Ground	If the protective ground connection is lost, all accessible conductive parts, including knobs and controls that may appear to be insulated, can render an electric shock.
Proper Power Cord	Use only the power cord and connector specified for the product and your operating locale. Use only a cord that is in good condition.
Proper Fuse	The fuse is mounted internally and is not considered user serviceable. The fuseholder accepts American sized fuses (1/4 in. dia.) or European sized fuses (5 mm dia). For 117V ac operation, the correct value is 1/2A, 250V ac, fast blowing (Bussman type AGC). For 230V ac operation, the correct value is 1/4A, 250V ac, slow blowing (Bussman type MDL or GDC) .
Operating Location	Do not operate this equipment under any of the following conditions: explosive atmospheres, in wet locations, in inclement weather, improper or unknown AC mains voltage, or if improperly fused.
Stay Out of the Box	To avoid personal injury (or worse), do not remove the product covers or panels. Do not operate the product without the covers and panels properly installed.
User-serviceable parts	There are no user serviceable parts inside the 602. In case of failure, refer all servicing to the factory. The complexity of the DSP circuitry as well as the special assembly tools required make the feasibility of field service doubtful.

2. Basics

If the particular combination of processors in the 602 is strange or foreign to you, then we suggest that you read and digest this section of the manual. If you should find some of the terminology strange, you'll find a glossary of terms at the end of the manual. A very good dictionary-style reference is also listed in the Bibliography.

2.1 What Does the 602 Do?

The 602 is a unique combination of four digital signal processors in one box: a versatile three-band parametric equalizer, a dynamic filter, a dynamics processor, and a digital delay. All of these processors are implemented in the digital domain and the 602 can accept (or output) signals in either the analog or digital domains.

One way to think about the particular combination of processors in the 602 is in terms of a modern mixing console. Today, most mixing consoles have microphone and line inputs, some sort of equalization, effects sends and returns, and (occasionally) on-board dynamics processing. For a typical voice-over session, you would probably have a compressor/limiter and a digital delay patched in as outboard processors. The 602 provides each of these processors wrapped into one tidy one rack-space package.

2.2 Digital and Analog Differences

A large difference between the 602 and a mixing console is that the processing functions in the 602 are implemented totally within the digital domain whereas those within the console are most likely implemented in the analog domain.

Outwardly there is no difference between an analog and a digital processor. A digital parametric equalizer has the same controls that you're familiar with in the analog world. Granted, the way that you access these controls may be different, but how much difference is there in seeing +9 dB on an LED display or in reading it off of a knob against a scale on the front panel?

2.3 Gain Setting

Wire is probably the only component of a sound system where we don't need to take signal levels into account (usually). Any other *active* component of a sound system that passes signal has a finite dynamic range. This means that our old friends dynamic range, headroom, and noise floor are present and must be accounted for.

Tackling these terms in reverse order, noise floor represents the signal level of the device's residual noise level. Realistically, this is somewhat lower than the lowest signal level that you'd want to process (unless you *want* the output to sound noisy).

Headroom is the difference between the average signal level and peak clipping. Peak clipping occurs because the processor can't increase its output to follow the signal. When this occurs, the output signal simply flat-tops over the period that it can't follow its input (sort of like clipping the tip of the peak off with a hedge-trimmer to level it off). Audibly speaking, clipping and using the hedge-trimmer are about equivalent.

Dynamic range is the difference between the highest signal that may pass (limited by peak clipping) and the lowest signal that will pass (limited by the noise floor). In a digital processor, a 0 dB signal may output a -120 dB noise floor but the smallest signal that may be represented by 18-bits is a -108 dB square wave (because there is only one bit to toggle on and off). Somewhere between these two points is the average level of your signal, as set by the processor's level control. Set the average level too high and peak clipping will smash your peaks flat, set it too low and suddenly the noise floor becomes audible (but you've got lots of headroom!).

The 602 allows you to set the signal levels in three different locations, which allows you to make the best tradeoff between headroom and dynamic range. The analog inputs each have gain controls to help you run these stages as hot as possible, without clipping. After conversion to digital form, some signals may be too hot for any signal processing that results in an increasing signal level. Thus, the 602 has a digital gain control that allows you to raise or lower the level sent to the digital processors. Finally, there is an overall digital output gain control allowing you to restore the signal level to "normal."

2.4 Equalization

Equalization is nothing more than selectively (or not) amplifying a signal based on frequency. Since audio signals consist of combinations of fundamental signals and their harmonics, changing the tonality or the spectral balance of a signal involves nothing more than altering the relationship of the fundamental to its harmonics, and of the harmonics to themselves. Each harmonic is responsible for one aspect of the audible character of a signal; knowing these relationships allow you to quickly zero-in on the correct frequency range of the signal and quickly apply boost or cut to enhance or correct what you are hearing.

The audio spectrum has several critical portions that are responsible for our perceptions of sounds that we hear:[1]

Range	Frequencies	Musical Location
Very Low Bass	16-64 Hz	1st and 2nd octaves.
Bass	64-256 Hz	3rd and 4th octaves.
Midrange	256-2048 Hz	5th, 6th, and 7th octaves.
Lisping Quality	3000 Hz	Between the 7th and 8th octaves.
Presence Range	4750-5000 Hz	Between the 8th and 9th octaves.
Brilliance	6500-16 kHz	Part of the 9th through the 10th octave.

2.4.1 Power and Fullness.

In the very low bass region lies the threshold of feeling, where the lowest sounds, like wind, room effects, and distant thunder, are felt, rather than heard. In the upper half of the first octave of this range, research has shown that the fundamentals of piano, organ and even the harp reach well into this range. Harvey Fletcher (of Fletcher-Munson fame) charted the sensitivity of the ear for various parts of the spectrum at levels that are lower than those of reality. Fletcher's compensation curves (the well known Fletcher-Munson curves) show that for equal loudness in this range at lower recorded and reproduced levels shows requirements for tremendous boosts, on the order of 10 to 30 dB. Aside from the subjective effects of this range, the ability to control unwanted sounds in this range is equally important to subdue stage rumble and outside traffic noise (especially important where there are subways beneath buildings!). Overemphasis caused by close cardioid microphone placement can cause muddiness in the overall sound; attenuating (cutting) the very-low-bass region can greatly improve overall clarity.

2.4.2 Rhythm and Musical Foundation.

In the bass region, most of the low, grave tones of the drum and piano can be found. Here we can also find the fundamentals of the rhythm section, as well as the foundation of all musical structure.

It was Leopold Stowkowski who said "If I had a thousand bass viols I could use them all!" This is not as extreme as it may sound. A bass viol, even though it is reinforced by its sounding board, generally plays single notes and possesses little dynamic range. In a large orchestra, as many as

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[1] The majority of the material in section 2.4 is taken from "Equalizing for Spectral Character," Langevin Corporation, 1966 Catalog.

eight bass viols may be used. A total of 1000 bass viols in this case would only give an additional 21 dB of level, which is not an inordinate amount given a glance at Mr. Fletcher's equal loudness curves. Pay attention to this range because the overall musical balance of your program can be controlled by equalizing or attenuating the 100 Hz range.

2.4.3 Telephone Quality

The ear is reasonably sensitive in the midrange frequencies, and sound restricted to this range has a telephone-like quality (which is generally why telephone-quality frequency response covers the 300-3 kHz range).

If you make the 6th octave (500-1024 Hz) louder with respect to the other octaves, the subjective result is a horn-like quality. If you emphasize the 7th octave (1000-2000 Hz), the effect is one of tinniness.

The fundamental tones in most music lie equally above and below middle C (261 Hz), from 128 to 512 Hz. As most instruments are rich in the first overtones, the majority of sound energy is found up to the 2.5 kHz range. Music editors and others engaged in listening to music over long periods find that listening fatigue can be reduced by attenuating the 5th, 6th, and 7th octaves by about 5 dB.

2.4.4 Lisping Quality

The 3 kHz range delivers a generous stimulus to the ear. At very loud levels the region of greatest ear sensitivity shifts downward from 5 kHz; this is why many "PA" speakers have broad peaks in this region. A characteristic of low-level signals peaked at 3 kHz is a "lisping" quality, and the total inability to distinguish labial sounds such as m, b, and v.

In wide-range lower level systems, a peak in the 3 kHz region has a masking effect on important recognition sounds, and on others which lie above 4 kHz. Brilliance and clarity are lost and without attenuation of this region, an unconscious strain with increasing fatigue is felt according to the amount of 3 kHz boost.

2.4.5 Presence Range

The usual band affecting clarity in male speech is 3000 to 6000 Hz. In a woman's voice, the fundamentals are roughly an octave higher than a man's, and a woman's range of consonant clarity lies between 5000 and 8000 Hz (the high-end of this range approaches a region of hearing insensitivity in humans). Furthermore, the total range of a woman's voice is about half that of a mans, stimulating fewer hearing nerves, and for this reason, is consequently still weaker upon reception.

Wide range sounds, especially those of singing voices, have fundamentals with harmonics in the 5 kHz region of good ear sensitivity. Voices that are powerful or rich with harmonics at 5 kHz sound especially pleasing, clear and full. Male opera singers are particularly favored with 5 kHz sounds, women less so. In popular music, this range shifts downward somewhat. It follows that voices deficient in the 5 kHz range can be enhanced in listening value by a generous boost on the order of 5 to 8 dB at 5 kHz. A secondary benefit of this boost is an apparent increase in level; a 6 dB rise at 5 kHz frequently gives an apparent increase of 3 dB to the overall signal.

Attenuating the 5 kHz range on instruments gives a "transparent" quality to the sound, providing, of course, that the remainder of the signal is otherwise wide range. Microphones having a dip in this region lack the "punch" or "presence" to which we (Americans) are accustomed.

2.4.6 Brilliance

Unvoiced consonants attributed to tooth, tongue and lip sounds are high in frequency, and reach the 10 kHz range. These frequencies account for some clarity and most brilliance, even though they contain less than 2% of the total speech energy. This also holds true for musical instruments; especially percussion. Boosting or cutting this range affects clarity and naturalness.

In speech, the 9th and 10th octaves impart intimacy although too much emphasis can make secondary speech sounds (lip smacking, etc.) objectionable (a good case for a downward expander).

Some microphones having a rise at the higher frequencies (especially omni microphones) benefit from some attenuation in this region. Those microphones having underdamped diaphragms may ring at these frequencies, causing an annoying sibilant distortion on speech. On musical forms using hand percussion, boosting this range frequently results in an astonishing and pleasing feeling of clarity.

2.4.7 Conclusions

When the article containing the above excerpts was written (probably around 1963), stereo was just becoming a commercial reality (you could still purchase mono and stereo versions of an LP and there were still more FM stations broadcasting in mono than stereo), and as many mixers contained rotary mix pots as those that used slide pots. The value of individual channel equalization was known, but it was both technologically and financially prohibitive. The article concludes thusly:

> *"With the advent of stereo and three-channel recording, nearly three times the equipment, with more elaboration, seems indicated, and expansion of console area in the horizontal plane offers the only direction in which to proceed. But a single engineer has arms only so long."*

How times have changed!

2.4.8 Equalizing for Speech

In broadcast, equalizers are often used to create a sonic personality for the station's on-air personalities. In the past, this has often meant using a single non-programmable equalizer in the announce mic's signal chain. Considering the inverse rule of the knobs (the more knobs you give them, the easier it is for someone to get hopelessly screwed up!), the attitude of most station's PDs and engineers was to hide the equalizer somewhere, preferably under lock and key. The 602makes it easy for each personality to have their own, individualized, curve. Granted, if you give the jocks access to the unit, someone will inevitably shoot themselves in the foot, but at least everyone can have their own curve.

Some general thoughts on speech equalization:

1. Try to use wider bandwidths. Narrower bandwidths (1/2 octave and less) are less audible (harder to hear) and are generally only useful for remedial work. Broader bandwidths are less obnoxious, more pleasing sounding, and easier to work with (especially if you're boosting a range of frequencies).
2. Try to avoid massive amounts of boost or cut. If you're only trying to impart a flavor (like sprinkling salt and pepper on a meal), then 6-8 dB of boost or cut should be all that you need.
3. A wide bandwidth cut is equivalent to a boost at the frequencies surrounding the cut.
4. A quick way to figure out what's going on is to set the level of one band of the equalizer to full boost (+18 dB), then switch to the frequency control and vary the frequency of that band of the equalizer while listening to program material fed through the unit. This usually makes quick work out of finding the region that you want to work on. Now reduce the level setting to something tasteful.

It's sometimes difficult to translate what you are hearing into the numbers that make equalizers happy. Seeing the frequencies associated with a voice or instrument can be helpful in deciding where equalization may be needed. The chart shown in Figure 2-1 shows the relationships of many different instruments, and a piano keyboard along with the frequencies involved.

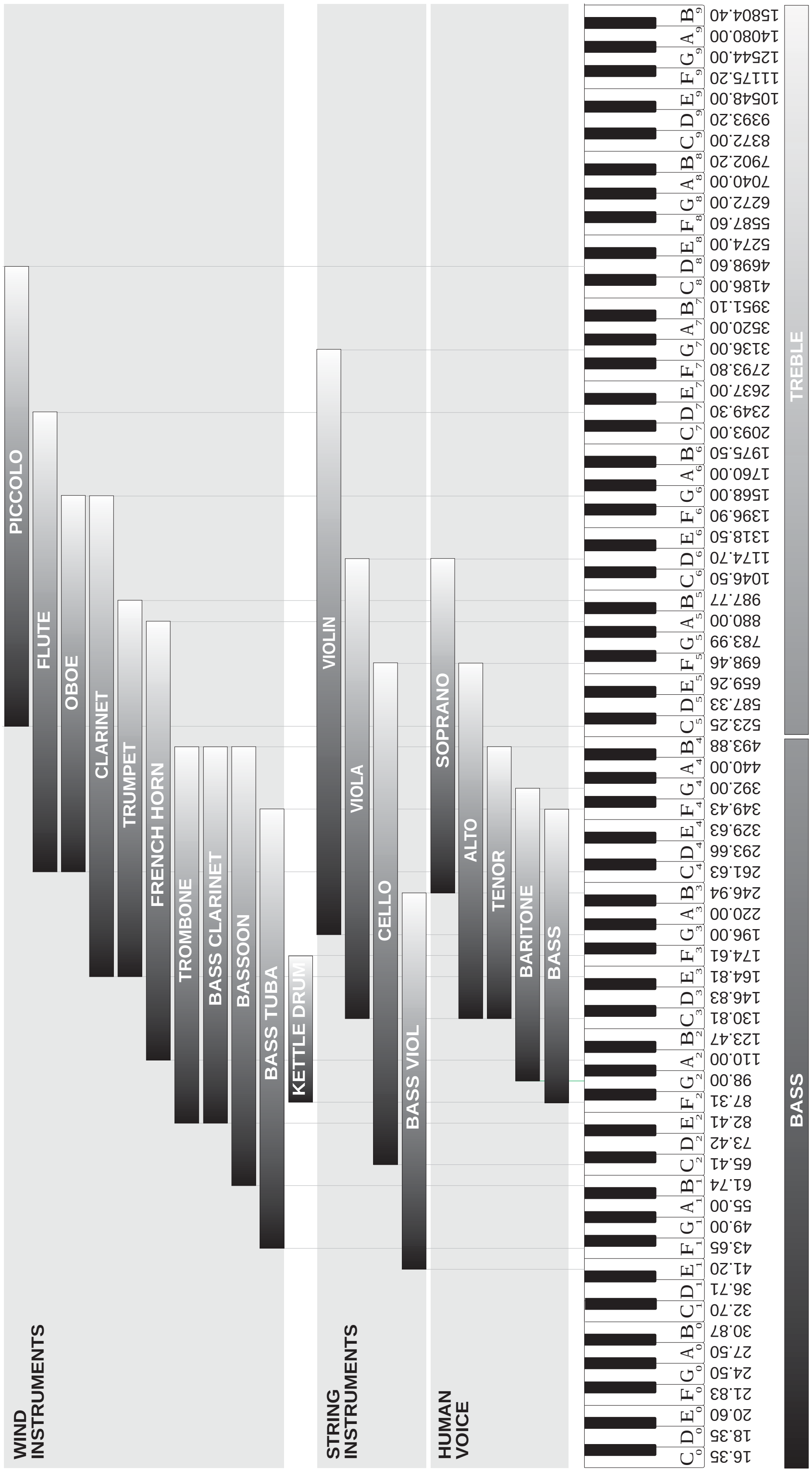

Figure 2-1. Relationships of Musical instruments, Piano, and actual frequencies.

2.4.9 Peaking or Shelving?

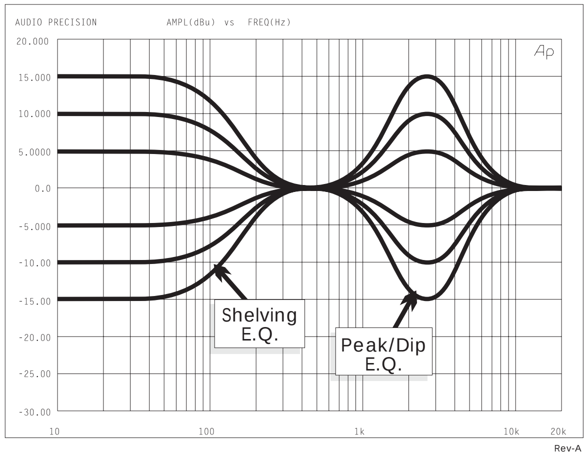

Figure 2-3. Shelving and peak/dip EQ curves.

The 602s equalizer can operate in either *peaking* or *shelving* mode. The two terms refer to the overall shape of the equalizer's frequency response curve. In Figure 2-3, you can see that the peaking equalizer's effect is concentrated at one frequency (the *center* frequency), with progressively less effect above or below the center frequency. The shelving equalizer (which acts more or less like the tone controls on a home stereo) affects frequencies above or below its characteristic frequency (depending on whether we're talking about a low-frequency shelving equalizer or a high-frequency shelving equalizer).

At very narrow bandwidths (small number), peaking equalizers exhibit a phenomenon known as "ringing." This quite aptly describes the effect of the equalizer being sharply resonant at its center frequency, which makes it *almost* oscillate.

In general, use the shelving curves to create overall color changes to the entire signal, and use the peaking curves to modify specific regions of the signal. The peaking curves bring another variable into play, "bandwidth" or "Q" as it is sometimes known. The bandwidth parameter simply tells you how much of the region surrounding the center frequency will be affected. Bandwidth and Q are inversely related; that is, a wide bandwidth (large number) corresponds to a low Q (small number).

2.5 De-Essing

De-essing is the process of removing "S" sounds from speech or singing. The technique was originally developed for motion picture dialogue recording when it was discovered that speech sounded more natural when the accentuation of sibilants ("s" sounds) was reduced. By sensing and limiting certain frequencies, the de-esser is intended to provide more specific control over some of the higher frequency vocal sounds that tend to become overemphasized.

Most sibilant vocal sounds like "s", "sh," and "t" are very difficult to reproduce electronically because they contain a large percentage of very high frequency harmonics. Since these sounds are so essential to the intelligibility of speech, they can't be simply removed with equalization. In fact, to help maintain articulation, many sound engineers routinely boost the higher frequencies of the vocal spectrum (3 kHz to 8 kHz), and/or use microphones with "presence curves" (like the Neumann U-87 or AKG C-414). However certain individuals and even certain languages contain overemphasized sibilants and any sort of high frequency boost only exacerbates the problem.

2.6 Noise Reduction

Noise reduction is the process of removing the noise from a signal without (hopefully) affecting the signal itself. There are two types of noise reduction: single ended (the 602), and double-ended (like Dolby noise reduction[2]).

A double-ended system such as the Dolby System eliminates noise contributed between its encode and decode processors. By necessity, this means that you must have access to the signal before it has noise added to it, and afterwards. For tape recorders and their ilk, this is perfect. Of course, if you feed a Dolby noise-reduced system a noisy signal, it will simply hand it back to you, without any added noise of course, but with just as much noise as you gave it to begin with (garbage in, garbage out or GIGO).

A single ended noise reduction system works on whatever signal you hand it. Single-ended systems depend on noise masking by the signal. That is, when the signal is present, it tends to mask the noise. So when the signal is quiet or absent, reduce the noise (by reducing the high-frequency response), and when the signal is present, remove the high-frequency rolloff and pray that the signal masks the noise.

If you're handed a noisy signal, then a single-ended noise reduction processor is your best weapon against the noise. If you combine this with some careful equalization, you'll probably end up with a signal that is more listenable.

2.7 Downward Expansion

Expansion is the process of increasing a signal's dynamic range, usually by increasing the signal's level by a precise amount for every dB over a magic signal level (the "threshold"). Unfortunately, this requires infinite (or at least *near* infinite) headroom.

A simple, but entirely satisfactory solution is to reduce the signal's level for every dB below a magic signal level (the "threshold"). This is called **downward expansion**. A similar and related device is the **signal gate**. You can think of a signal gate as a special case of a downward expander (or vice-versa if you must). Both devices reduce their output when their input signal falls below threshold. The difference is the rate (not speed) at which they do it. The 602’s downward expander output falls at an adjustable rate for every 1 dB below threshold of the input signal. A gate's output falls by a nearly infinite amount for the slightest change, below threshold, of the input signal. You can think of a gate as a downward expander taken to the extreme, or you can think of a downward expander as a subtle example of a gate.

Gates are generally used to remove leakage (unwanted signals from nearby sources) from a signal. Downward expanders are used to remove extraneous noise and to increase dynamic range by making the softer parts softer.

Compressors or limiters (for the purposes of this discussion, a limiter is simply a high-ratio compressor) are often used to reduce dynamic range by setting an upper limit on larger signals. Sometimes, when you're trying to fit a signal through a transmission channel, it's better to put processing to work on the lower end of the dynamic range than on the upper end. In other words, instead of reducing the amount of change at the upper end of the dynamic range with a compressor or limiter, try using a downward expander to increase the amount of change at the lower end of the dynamic range.

2.8 Compression

Many times, a signal's dynamic range must be modified to allow it to pass through a transmission channel without clipping or becoming noisy. Most often, audio engineers patch in a compressor to restrict the dynamic range of a signal.

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2 Dolby is a trademark of Dolby Laboratories, San Francisco, CA, USA.

A compressor is a gain control device whose output is nearly constant in spite of variations in its input level. A simple analogy is: you're holding the volume control for a sound system and you're told to turn it down if the level gets louder than it is now and turn it down enough that the level is the same as it is now. Figure 2-4 illustrates this concept graphically.

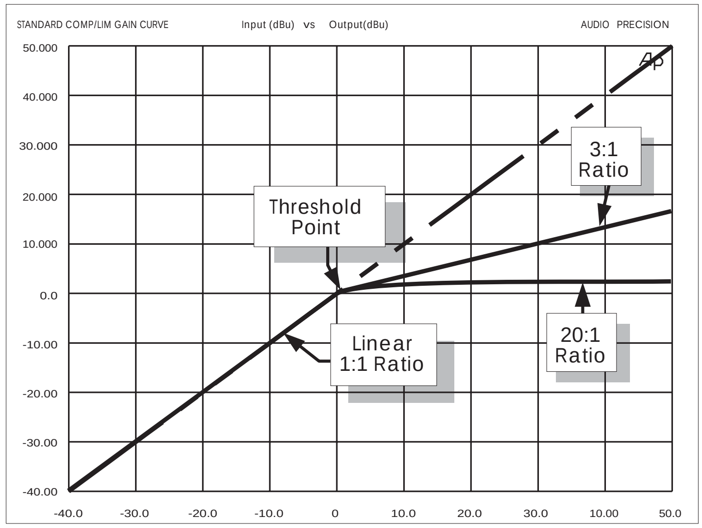

Figure 2-4. Input/Output curves of a compressor.

Compressors can also be used creatively, that is to create an effect. In this case, the "rules" such as they are, go out of the window. A large amount of compression applied to a voice-over can create the impression of excitement or intimacy, or simply help make the signal very loud in a controlled manner, which might be useful in ensuring that the voice is **always** heard.

For most voice applications, pick a moderate ratio (4:1 to 8:1), and set the threshold low enough to achieve 6 dB or so of gain reduction. Set the attack time to retain some of the "edge" of each word, and set the release time fast enough to follow the speech. For a heavily limited sound, set the ratio higher (10:1 or higher) and use very-fast release times[3].

For musical applications, use low ratios (1.5:1 to 6:1) unless you want a deliberately squashed sound. Set the threshold to achieve 4 to 6 dB of gain reduction. This setting is useful for subtly controlling occasional peaks. To prevent peak overload of a subsequent device, use the highest ratio and set the threshold to achieve 2 to 4 dB gain reduction on the highest peaks. If you’re using the 602 to ride gain on mixed program, consider using the AGC section.

2.9 AGC

The letters AGC stand for Automatic Gain Control. An AGC can also be considered as a special case of a compressor having a relatively low ratio (1:1 -> 4:1) and a very low threshold level and a gated release time. Thus, any signal that exceeds the threshold causes some degree of gain reduction. Additional gain, applied after the compressor, brings the signal level back up to line level.

Functionally, an AGC works like an invisible (and hopefully inaudible) operator who monitors the audio level and imperceptibly raises or lowers the gain to maintain the audio level at some predetermined point. In its most simplistic form, that's all there is to it. (But there is a BUT, as you will see.)

AGC amplifiers have been with us for many years. In the broadcast world, the old Gates Level-Devil and CBS Labs Audimax are both examples of old (circa 1960) products that performed this function. The feature that sets these guys apart from common-ordinary-garden-variety compressors is: gated, program-controlled release.

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[3] Fast attack times may become audible because of the time required to compute the amount of gain reduction (analog compressors have the same problem, but they usually limit the minimum attack time so you never have the problem). You can use the 602's compressor look-ahead parameter to "buy some time" so that the initial overshoot of the signal is controlled by the compressor. Fast release times cause problems of their own because changes in the gain reduction may occur during a single cycle of the waveform, causing distortion. Again, analog compressors are not immune to this problem either.

If you remember back to the early days of TV, remember when someone at the network screwed up and let the program lapse, the compressor at the local station would release and whoooosh, up would come the noise floor...until the guy at network woke up, in which case it went suuuuuuuck and back into the program audio. Both the Level-Devil and Audimax fixed this problem by making the release time of the compressor a function of the program audio. That is, they inhibited the gain reduction release if there was no audio present. If audio was present, then the compressor was free to release as much as it wanted, but if there was no audio present, the unit remained at the amount of gain reduction in force before the audio loss.

Both the Audimax and the Level-Devil depended upon silence to control the gain-release function. In practice, the silence detector can be fooled by a noisy input signal. Since the AGC/Leveler needs to work at very low threshold levels in order to accommodate a wide range of input levels (ideally, you want the AGC/Leveler to function with signals ranging from near thermal noise to high line level), an ordinary signal-present detector would respond to hum or noise by mistaking it for a valid signal.

If you try using a simple compressor as an AGC, there is no signal-controlled gated-release function. Thus the overall gain is highest anytime that the signal falls below the compressor's threshold. By itself, this isn't disastrous (perfectly workable with a noiseless input signal), but the sudden change in noise level when a normal-level signal presents itself is a dead giveaway that your compressor is lacking in the IQ department. This is the BUT mentioned earlier.

The 602's AGC function performs some analysis on the signal in order to make an informed decision about the signal's nature. If the signal is determined to be noise or silence, then the AGC's release function is inhibited. When the signal analyzer detects that the signal has returned, the AGC is again allowed to release, which causes the gain to rise or fall in response to the signal level.

2.10 Delay

One of the simplest things that you can do to an audio signal to dramatically change its character is to add in a delayed version of the signal. By adjusting parameters such as delay time, delay level, and feedback, you can create impressions of time, space, distance or reflection. The delay in the 602 is a two-channel delay with paralleled inputs and separated outputs. The feedback paths between input and output are cross-coupled. This means that delay one's output feeds delay two's input and delay two's output feeds delay one's input.

For example, you can get a '50s sound by adding 250 ms delay to a signal. Adding feedback makes the delay effect linger, since the feedback causes the echoes to repeat until they die out. By shortening the delay and fiddling with the feedback, you can simulate reflective rooms of various dimensions. Making the delay times slightly different "spreads" the sound, eliminating the point-source effect. If the delay times are long enough, you'll hear the echoes bouncing back and forth between the speakers.

Note: the dual-delay used in the 602 is not sufficient to create any sort of realistic reverb. Good sounding reverberation requires a multi-tapped delay line.

2.11 Modulated Delay

Yet another wrinkle on delay is modulating the delay time. This causes the delay time to vary according to the frequency and amplitude of the modulating signal. If you choose a short delay time (typically around 1 ms), a low modulation frequency, and roughly a 50-50 mix of direct signal and delayed signal, you'll get flanging. Adding feedback accentuates the effect.

If you alter the mix to favor the delayed signal and raise the modulation frequency, you'll get pitch bending, vibrato, or chorusing.

2.12 MIDI

If you aren't aware of MIDI, well... Several years ago, a number of musical synthesizer manufacturers somehow agreed on a serial data protocol to exchange control information between synthesizers. They called the result MIDI: Musical Instrument Digital Interface. The success of this standard is phenomenal (not that it is perfect) and the ability to control something via MIDI has been applied to everything from synthesizers to signal processors to lighting systems.

Nearly every parameter of the 602 may be controlled or modified via MIDI. The 602's MIDI implementation is described in Appendix C.

What does this mean for the 602? At a very basic level, it means that you could have several setups stored in the 602 and change between them remotely. If you're a broadcaster, you could have a MIDI controller output program change commands based on a clock, which would change the settings of the on-air 602 to the personalized settings for each of your on-air personalities. If you're a musician, it means that the settings of the 602 can be changed from note to note, measure to measure, during the solo, between songs, ...get the picture? The 602's MIDI capability can also be used for dynamic parameter control (realtime control via MIDI continuous controller) and parameter editing (making parameter changes via MIDI). Both of these activities require an external MIDI controller or a MIDI-equipped computer.

2.13 Program Memory

The 602 has 256 memory locations for program storage. The first 128 locations are reserved for user memory; the last 128 locations are reserved for the factory supplied programs. You may recall any factory program (or any other stored program), edit it (modify any of its parameters), and store the result into one of the user memory locations. Later, these programs may be recalled via the front panel, or via MIDI for reuse or further editing.

3. Technical Tutorial

This section discusses a multitude of things, all related to getting signals in and out of the 602.

3.1 Matching Levels vs Matching Impedances

In any audio equipment application, the question of "matching" inevitably comes up. Without digging a hole any deeper than absolutely necessary, we offer the following discussion to (hopefully) clarify your understanding of the subject.

Over the years, we have all had impedance matching pounded into our heads. This is important only for ancient audio systems, power amplifiers, and RF. Technically speaking, the reason is power transfer, which reaches a maximum when source and load are matched. Modern audio systems are voltage transmission systems and source and load matching is not only unnecessary, but undesirable as well.

- ❒ Ancient audio systems operate at 600 ohms (or some other impedance value), and must be matched, both at their inputs and at their outputs. Generally speaking, if you are dealing with equipment that uses vacuum tubes, or was designed prior to 1970, you should be concerned about matching. These units were designed when audio systems were based on maximum power transfer, hence the need for input/output matching.
- ❒ Power amplifiers are fussy because an abnormally low load impedance generally means a visit to the amp hospital. Thus, it's important to know what the total impedance of the pile of speakers connected to the amplifier really is.
- ❒ RF systems are matched because we really are concerned with maximum power transfer and with matching the impedance of the transmission line (keeps nasty things from happening). Video signals (composite, baseband, or otherwise) should be treated like RF.

Some folks seem to believe that balanced/unbalanced lines and impedances are related; or even worse that they are associated with a particular type of connector. Not so. Unbalanced signals are not necessarily high-impedance and balanced signals/lines are not necessarily low-impedance. Similarly, although 1/4 inch jacks are typically used for things like guitars (which are high-impedance and unbalanced), this does not predispose them to only this usage. After all, 1/4 inch jacks are sometimes used for loudspeakers, which are anything but high-impedance. Therefore, the presence of 3-pin XLR connectors should not be construed to mean that the input or output is low-impedance (or high-impedance). The same applies to 1/4 inch jacks.

So, what is really important? Signal level, and (to a much lesser degree), the impedance relation between an output (signal source) and the input that it connects to (signal receiver).

Signal level is very important. Mismatch causes either loss of headroom or loss of signal-to-noise ratio. Thus, microphone inputs should only see signals originating from a microphone, a direct (DI) box, or an output designated microphone-level output. Electrically, this is in the range of approximately -70 to -20 dBm. Line inputs should only see signals in the -10 to +24 dBm/dBu range. Guitars, high-impedance microphones, and many electronic keyboards do not qualify as line-level sources.

The impedance relation between outputs and inputs needs to be considered, but only in the following way:

> **Always make sure that a device's input impedance is higher than the output source impedance of the device that drives it.**

Some manufacturers state a relatively high-impedance figure as the output impedance of their equipment. What they really mean is that this is the minimum load impedance that they would like their gear to see. In most cases, seeing a output impedance figure of 10,000 (10K) ohms or higher from modern equipment that requires power (batteries or AC) is an instance of this type

of rating. If so, then the input impedance of the succeeding input must be equal to or greater than the output impedance of the driving device.

Symetrix equipment inputs are designed to bridge (be greater than 10 times the actual source impedance) the output of whatever device drives the input. Symetrix equipment outputs are designed to drive 600-ohm or higher loads (600-ohm loads are an archaic practice that won't go away). You don't need to terminate the output with a 600-ohm resistor if you aren't driving a 600-ohm load. If you don't understand the concept of termination, you probably don't need to anyway.

The two facts that you need to derive from this discussion are:

1. Match signal levels for best headroom and signal-to-noise ratio.

2. For audio, impedance matching is only needed for antique equipment and power amplifier outputs. In all other cases, ensure that your inputs bridge (are in the range of 2 to 200 times the output source impedance) your outputs.

3.2 Signal Levels

The 602 is designed around studio/professional line levels: +4 dBu or 1.23 volts. The unit is quiet enough to operate at lower signal levels such as those found in semi-pro or musical-instrument (MI) equipment (-10 dBu or 300 millivolts).

3.3 I/O Impedances

The 602 is designed to interface into almost any recording studio or sound reinforcement application. This includes:

❒ 600-ohm systems where input and output impedances are matched.

❒ Unbalanced semi-professional equipment applications.

❒ Modern bridging systems where inputs bridge and outputs are low source impedances (voltage transmission systems).

The 602's input impedance is 12.5 kΩ balanced, and 9.4 kΩ unbalanced. The inputs may be driven from any source (balanced or unbalanced) capable of delivering at least -10 dBu into the aforementioned impedances.

The 602's output impedance is 300-ohms balanced, 150-ohms unbalanced. The output line driver delivers +21.5 dBm into 600-ohm balanced loads or +15.5 dBm into 600-ohm unbalanced loads.

3.4 Polarity Convention

The 602 uses the international standard polarity convention of pin 2 hot. Therefore:

XLR	Tip-Ring-Sleeve	Signal
1	Sleeve	Ground
2	Tip	High
3	Ring	Low

If your system uses balanced inputs and outputs, and uses the 602 this way, then the polarity convention is unimportant. If your system is both balanced and unbalanced, then you must pay attention to this, especially when going in and coming out through different connector types (like input on an XLR, output on a phone jack).

3.5 Input and Output Connections

Figure 3-1 illustrates how to connect the 602 to balanced and unbalanced analog sources and analog loads.

To operate the 602 from unbalanced sources, run a 2-conductor shielded cable (that's two conductors plus the shield) from the source to the 602. At the source, connect the low/minus side to the shield, these connect to the source's ground; connect the high/plus side to the source's signal connection. At the 602, the high/plus wire connects to pin 2, the low/minus wire connects to pin 3, and the shield (always) connects to pin 1. This is the preferred method as it makes best use of the 602's balanced input (even though the source is unbalanced). The other alternative shown in Figure 3-1 converts the 602's balanced input into an unbalanced input at the input connector. This works, but is more susceptible to hum and buzz than the preferred method. There is no level difference between either method.

You can drive unbalanced loads with the 602's outputs by using the XLR connector with pin 3 left open. In an emergency (the show must go on), you can ground pin 3, but if you have the choice...leave it open. If you must ground pin 3, it is must be grounded at the 602, rather than at the other end of the cable. The price, regardless of whether or not pin 3 is grounded is 6 dB less output level. This can be easily made up via the output gain controls. If your system is wired with pin 3 hot, **pin 2 must float** if you are driving an unbalanced load.

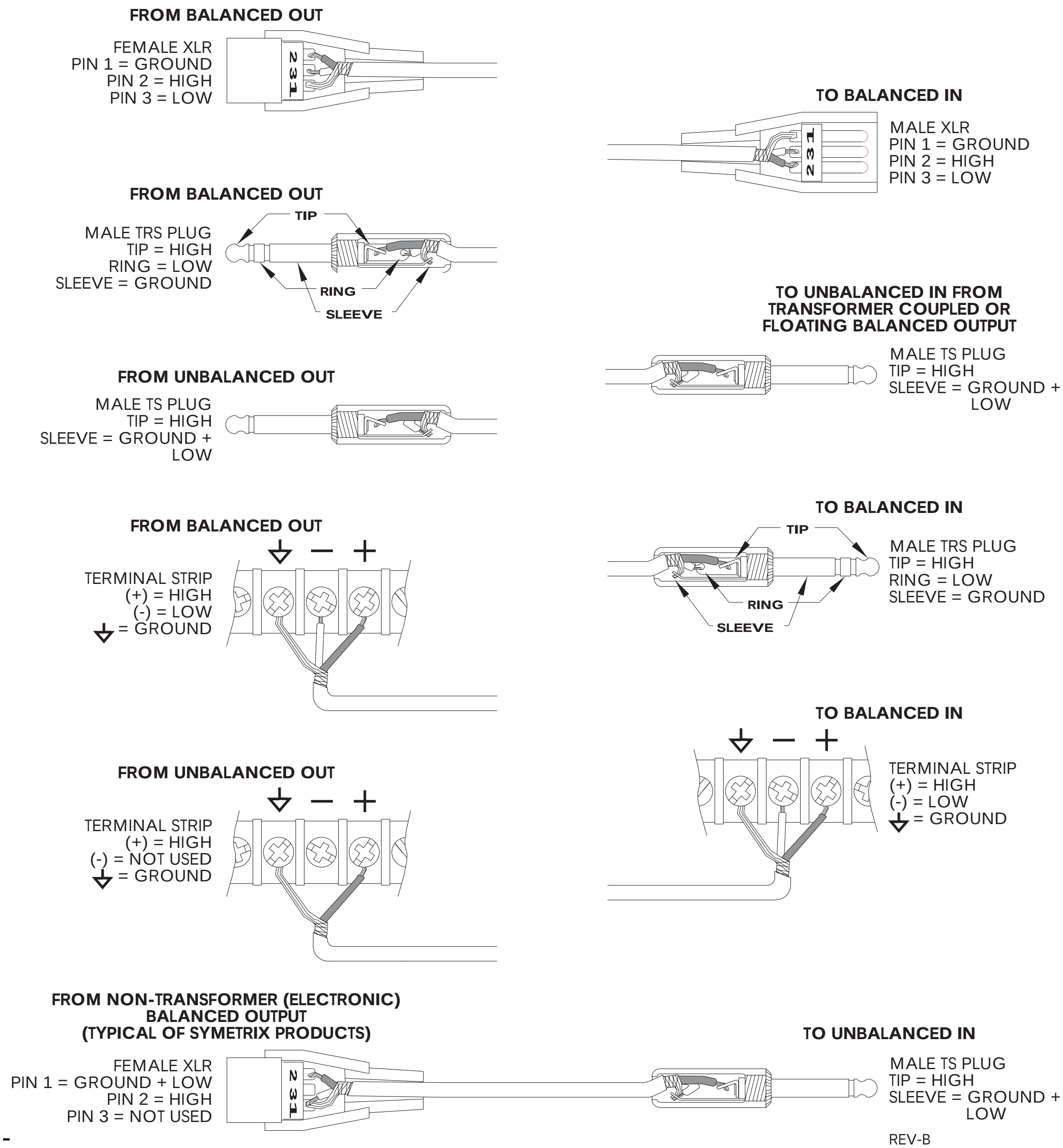


Figure 3-1. Input and output connector wiring. These diagrams represent the majority of connectors used in modern audio equipment. Locate the source connector in the left column and match it up with the destination connector in the right column. Wire your cable according to the diagrams

3.6 Digital I/O Considerations

The 602 has two similar, but different, digital input/output formats: AES/EBU and S/PDIF. The AES/EBU format uses XLR connectors, is balanced, and operates at 110 ohms line impedance. The S/PDIF (Sony-Phillips Digital InterFace) format uses RCA connectors, is unbalanced, and operates at 75 ohms line impedance. The digital input and output is transformer coupled for freedom from ground loops.

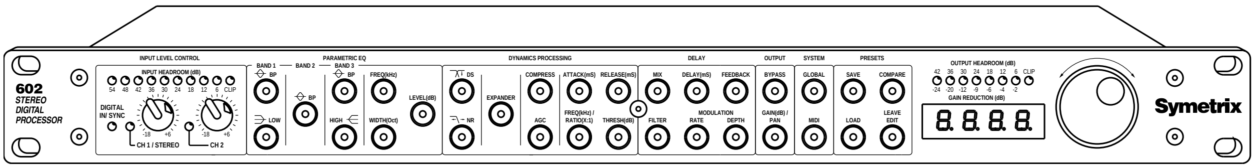
3.7 MIDI I/O Considerations

The 602 has two MIDI connections: MIDI in and MIDI out. There is no MIDI thru connection. Both connectors follow the standard defined by the IMA (International MIDI Association).

The MIDI out jack echoes the MIDI input, minus any sysex messages aimed at the 602 that owns the MIDI connectors. All other messages are echoed. There is a small throughput delay that may become significant if many 602s are wired in cascade (series).

Notes

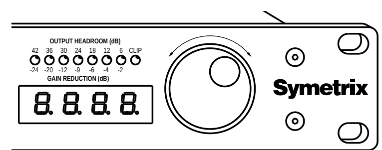
4. Front Panel Overview



4.1 User Interface Summary

The user interface of the 602 has been designed to be powerful yet intuitive. Most switches have only one function and where a switch has several functions, the display prompts for the parameter in question. There are no hierarchical menus.

The parameter adjustment wheel (Wheel) modifies the parameter or function selected via the front panel switches. The Wheel is sensitive to direction, and velocity. Thus, turning the Wheel quickly causes the display to change very quickly, and turning the Wheel slowly causes the display to change very slowly (as if there were a gear reduction unit on the Wheel).

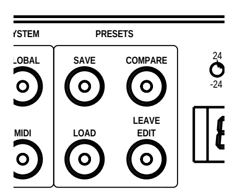


Select a function for editing by pressing its associated switch once. The switch begins flashing. Next, select the parameter that you want to edit and press its switch. The display indicates the current value. Turning the Wheel changes the value.

If a function switch has several choices, the display indicates the choice, and the Wheel cycles through the options. An example of this sort of function switch is the GLOBAL switch.

4.1.1 Loading Programs

A program is nothing more than a group of control settings. To load (recall) a program, press the LEAVE EDIT switch to return to the top-most control level. The display now indicates the current program number. Rotate the Wheel until the desired program number appears flashing in the display. Press the LOAD switch. When the program has loaded, the display says **donE** and the number stops flashing.



Note: the 602 always loads a **copy** of the program stored in program memory into the edit buffer. The contents of the edit buffer are lost only when another program has been loaded. If the program stored in the edit buffer is 'dirty' (i.e. it has been modified), the red SAVE switch flashes. It is possible to intentionally overwrite the current program by holding down the LOAD switch, regardless of its save status.

4.1.2 Saving Programs

To save a program, press the LEAVE EDIT switch to return to the top-most control level. The red SAVE switch should be flashing (if not, then the program in the edit buffer has not been modified; it doesn't need to be saved). Turn the Wheel to select the desired save location (memory locations 1 through 128), press and hold the SAVE switch until the display says **donE**. Program numbers above 128 are read-only. The save operation displays **- -** if you try to save a program to one of these numbers.

If you press and hold the SAVE switch without pressing LEAVE EDIT first, the 602 performs the save operation (using the current program number) and returns you to where you were.

Caution

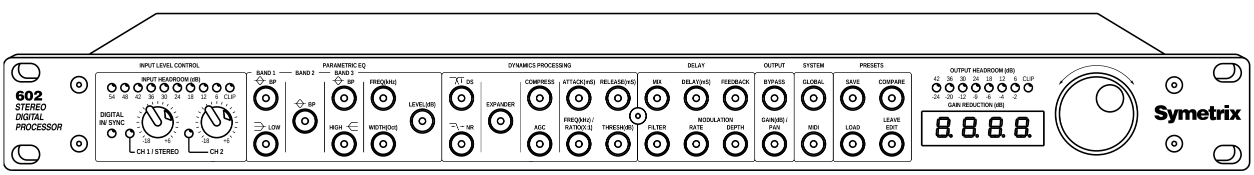
If the LOAD switch is flashing, this indicates that the current program number is different than the program number that the edit buffer was loaded from. If you press and hold the SAVE switch, you will overwrite the program number that is visible when you press LEAVE EDIT.

4.1.3 Comparing Programs

You can compare the program in the edit buffer with the unedited version of the program. Pressing the COMPARE switch toggles the 602 between the edited and unedited versions of the program. It is not possible to compare the edited program with any other program. The display toggles between **OLd** and **CUrr**ent to remind you what you're listening to. Pressing any parameter switch instantly returns you (and the outputs) to **CUrr**.

4.2 Rate of Change Parameter

In addition to the parameters visible on the front panel, many of the 602's parameters have a rate-of-change parameter (**rt**) associated with them. The rate parameter affects how quickly the parameter changes from its current value to its new value, either under direction of the front panel or MIDI. In essence, the rate parameter (rt) affects how fast the knob can turn (the knob is a "virtual knob" that represents a parameter that can be adjusted using the Wheel or via MIDI). You can see which parameters have an associated rate parameter by referring to the table in section C.3.1.



4.3 Input Level Control Block

This switch and control block sets the operating conditions for the analog-inputs of the 602. The controls and indicators operate as follows:

INPUT HEADROOM (DB) — LED display indicates amount of headroom remaining at the output of the A-D converters in the 602. The display ballistics are peak reading; the display should be interpreted as the absolute amount of headroom remaining.

The numbered part of the display reads the higest peak **digital** signal level of the two input channels (left and right) after the digital gain control at the input to the DSP section. The display can (temporarily) indicate the signal level of the left or right DSP channels via the GLOBAL switch.

The Clip LED responds only to overload at the output of the analog line input amplifiers. To maximize the dynamic range, set either of the two input gain controls (Ch 1/Stereo or Ch 2) so that the green 2 dB LED illuminates. The red CLIP LED should never illuminate.

DIGITAL IN/SYNC

The DIGITAL IN/SYNC LED indicates the presence of digital signals at either the AES/EBU or S/PDIF digital inputs. This LED also flashes to indicate error conditions (including no signal present) occurring with either of the digital inputs.

CH1/Stereo LED

This LED indicates that the CH1/Stereo gain control is active.

CH1

This rotary control determines the gain of the channel 1 line input circuit. In stereo mode, the ch1 and ch2 LEDs indicate whether the gain controls are separated or ganged.

CH2 LED

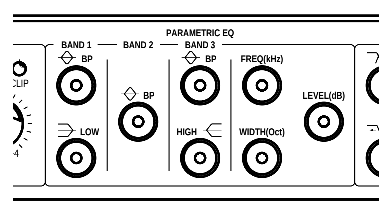
This LED indicates that the CH2 control is active.

CH2

This rotary control determines the gain of the channel 2 line input circuit.

The three status LEDs indicate the status of their associated input as well as indicating error conditions. For the analog inputs, the status LEDs also indicate which gain control is active. If the DIGITAL IN/SYNC LED flashes, this indicates the loss of digital data at the digital input (if a mode requiring either digital data or digital clock has been selected).

4.4 Parametric EQ Block



The parametric EQ block encompasses a full-function three-band parametric equalizer. All three bands provide reciprocal peak/dip equalization and bands 1 and 3 may be individually switched to shelving curves. For flexibility, each equalizer band covers the entire frequency range. The rate of change for level and frequency (bandwidth and center frequency) may be altered by holding down the LEVEL or FREQ switch until the display shows **rt**.

4.4.1 EQ Band Select

BAND 1

Band 1 is a bandpass peak/dip or low-frequency shelf. The EQ range is +18 dB to -50 dB, 31 Hz to 21.11 kHz, .05 to 3-octaves bandwidth (Q = 29 to 0.4). Pressing this switch toggles the Band 1 equalizer between in and **out**.

BAND 2

Band 2 is a bandpass peak/dip. The EQ range is +18 dB to -50 dB, 31 Hz to 21.11 kHz, .05 to 3-octaves bandwidth (Q = 29 to 0.4). Pressing this switch toggles the Band 2 equalizer between in and **out**.

BAND 3

Band 3 is a bandpass peak/dip or high-frequency shelf. The EQ range is +18 dB to -50 dB, 31 Hz to 21.11 kHz, .05 to 3-octaves bandwidth (Q = 29 to 0.4). Pressing this switch toggles the Band 3 equalizer between in and **out**.

4.4.2 EQ Parameter Group

The switches in the parameter group modify the settings of the selected (flashing) parametric EQ section. Each switch has the following action:

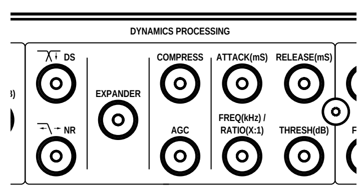
FREQ (KHZ)

The Wheel modifies the center frequency of the selected equalizer from 31 Hz to 21.11 kHz. Holding down the FREQ switch allows setting the time constant for the rate of change of the center frequency. Shares time constant (**rt**) with WIDTH switch. The filter frequencies are 1/10th octave ISO standard frequencies except for some special power line harmonic frequencies at the low end.

LEVEL (DB)

The Wheel modifies the amount of boost or cut from +18 dB to -50 dB.

The display indicates the filter's boost/cut setting in dB. The display reads **out** when no filter contribution has been set (same as LEVEL at 0/out). Holding down the LEVEL switch allows setting the time constant for the rate of change; the display reads **rt**. This affects how quickly the LEVEL setting changes either due to MIDI command, program change, or rotation of the Wheel.



WIDTH (OCT)

The Wheel modifies the bandwidth from 0.05-octaves (very sharp or narrow) to 3-octaves (quite broad). The rate-of-change value is shared with and accessed with the FREQ switch since both are frequency-related parameters.

4.5 Dynamics Processing Block

The dynamics processing block encompasses the de-esser, dynamic noise reduction (NR), downard-expander, compressor, and AGC. Whenever one of the dynamics processors has been selected for edting, the output LED display changes to a gain-reduction display

Note: All of the dynamics blocks use a threshold parameter. Unlike analog processors that you may be familiar with, each of the threshold settings in the 602 reference to digital clipping (full-scale) rather than to some nominal signal level (like 0 dBu). This means that

you may not be able to directly translate threshold settings that you are familiar with from the analog world to the digital world.

4.5.1 Dynamic Noise Reduction Block

The dynamic noise reduction (NR) block uses a variable frequency low-pass filter to perform single-ended noise reduction. The NR block is a feedback system; the amount that the filter 'opens up' is self-limiting and dependent on the high-frequency content of the input signal and the THRESHold setting. At higher THRESHold settings, there will always be some high-frequency loss.

NR Switch	Toggles the NR between active and **out**. When active, the FREQ and THRESH switches are active. When editing, the NR switch LED flashes, otherwise it reflects the state of the NR (in or out). Whenever NR is currently being edited, the output headroom LED display changes to indicate gain-reduction.
FREQ Switch	Sets the resting frequency of the NR (the -3 dB point of the dynamic lowpass filter when there is no input signal).
THRESHOLD Switch	The NR uses two threshold settings, one relative (display reads **r**), and the other absolute (the display reads **A**). You access the two threshold settings by pressing the THRESH switch when the NR has been selected. The NR reacts to the ratio of the signal passing through the adaptive lowpass filter and the signal being rejected by the adaptive lowpass filter. Higher (less negative) relative threshold settings require larger amounts of high-frequency content to cause the filter to 'open up.' The absolute threshold setting determines the transition point below which the NR system ignores the high-frequency content and relies strictly upon signal level information. Typically the absolute threshold should be set to equal the noise floor of the program material; the useful range for this parameter being from -80 to -50 dB.

4.5.2 De-Esser Block

Like the NR system, the de-esser is a feedback control system. The de-esser uses a broadband limiter with a peaked highpass filter in its sidechain; the frequency response is always flat regardless of the degree of de-essing. The attack, release, and frequency switches are functional and these parameters are also accessible via MIDI or the realtime editor.

DS Switch	Toggles the de-esser between active and **out**. When editing, the de-esser switch LED flashes, otherwise it reflects the state of the de-esser (in or out). Whenever editing the de-esser parameters, the output LED display changes to read gain-reduction.

THRESHOLD Switch

Sets the relative threshold for the start of de-ess action. The de-esser measures the energy on each side (highpass and lowpass) of the sidechain filter. Sibilant sounds above this threshold level are reduced in level. The absolute threshold (adjustable via the realtime editor) sets a minimum level that the signal must exceed to receive de-essing.(see Appendix A).

ATTACK Switch

Sets the time required for the de-esser to engage. This means that the input signal must remain above the THRESHold setting for a time that is longer than the attack time. The ATTACK time ranges from 0.1ms (100 microseconds) to 10,000 ms (the display reads 9999, but the time is really 10,000 milliseconds or 10 seconds).

RELEASE Switch

Sets the time required for the de-esser to recover once the sibilant sound has ceased. The time displayed is the time required for full decay in response to a large, above-THRESHold change in the input signal. The RELEASE time ranges from 100 ms (100 milliseconds) to 10,000 ms (the display reads 9999, but the time is really 10,000 milliseconds or 10 seconds).

FREQ Switch

Sets the transition frequency of the de-esser's sidechain filter. The frequency can be varied from 31 Hz to 21,112 Hz. The default frequency is 5 kHz.

4.5.3 Downward Expander Block

The downward expander reduces its gain for any signal level below the THRESHold setting.

EXPANDER Switch

Toggles the Expander between in and **out**. When editing, the expander switch LED flashes, otherwise it reflects the state of the expander (in or out). When active, all of the dynamics parameter modification switches are active. Whenever the expander is currently being edited, the output LED display changes to read gain-reduction.

ATTACK Switch

Sets the time required for the expander to terminate expansion. This means that the input signal must remain above the THRESHold setting for a time that is longer than the attack time. The ATTACK time ranges from 0.1ms (100 microseconds) to 10,000 ms (the display reads 9999, but the time is really 10,000 milliseconds or 10 seconds).

RELEASE Switch	Sets the time required for the expander's gain to decay once the input signal has fallen below threshold. The time displayed is the time required for full decay in response to a large, below-THRESHold change in the input signal. The RELEASE time ranges from 100 ms (100 milliseconds) to 10,000 ms (the display reads 9999, but the time is really 10,000 milliseconds or 10 seconds).
RATIO Switch	Sets the expansion gain RATIO (expansion ratio). The range is from 1.0 (out) to 8 (ratio of 1:8, or 1 dB input change to 8 dB output change).
THRESHOLD Switch	Sets the THRESHold for start of expansion. Signals below this level are reduced in level by an amount dependent on the setting of the expansion gain RATIO , and the difference between the threshold setting and the actual signal level.

4.5.4 Compressor Parameter Block

The compressor reduces its gain for any signal level above the threshold setting. The COMPRESSOR switch's LED indicates that the compressor is active. When editing, the compressor switch LED flashes, otherwise it reflects the state of the compressor (in or out). Whenever editing the compressor or AGC parameters, the output LED display changes to read gain-reduction. The compressor block and the AGC block are mutually exclusive; you can only use one of them at a time.

There is no output gain control; the 602 computes the correct amount of makeup gain based on the threshold and ratio settings (although the auto-makeup gain feature can be defeated and the amount of makeup gain can be set manually). The shape of the knee of the gain-reduction curve can be adjusted via MIDI or the realtime editor (see Appendix A).

ATTACK Switch	Adjusts the ATTACK time (milliseconds) of the compressor (time required for an above-THRESHold signal to cause gain reduction).
RELEASE Switch	Adjusts the release time (time, in milliseconds), required for the gain to return to the below-threshold value.
RATIO Switch	Controls the compression gain ratio (compression ratio). The range is from 1:1 (out) to 10:1. A 10:1 ratio means that a 10 dB input change results in a 1 dB output change (provided that the level of the entire change was above the threshold setting). The compressor's makup gain may be set manually by pressing the RATIO switch until the display reads **gAIn**. Set the makeup gain using the Wheel.
THRESHOLD Switch	Sets threshold for the start of compression. Signals above this level are reduced in level by an amount dependent on the setting of the compression ratio and the difference between the threshold setting and the actual signal level.

4.5.5 AGC Block

The AGC is a variation on a compressor that operates over a wide range of signal levels while trying to keep its output level constant. The AGC switch LED indicates that the AGC is active. When editing, the AGC switch LED flashes, otherwise it reflects the state of the AGC (in or out). Whenever editing the AGC parameters, the output LED display changes to read gain-reduction. The AGC block and the compressor block are mutually exclusive; you can only use one of them at a time.

There is no output gain control; the 602 computes the correct amount of makeup gain based on the threshold and ratio settings (although the auto-makeup gain feature can be defeated and the amount of makeup gain can be set manually). The shape of the knee of the gain-reduction curve can be adjusted via MIDI or the realtime editor (see Appendix A).

AGC Switch	The AGC switch LED indicates that the AGC/Leveler is active. When editing, the AGC switch LED flashes, otherwise it reflects the state of the AGC (in or out). There is no output gain control; the 602 computes the correct amount of makeup gain based on the ratio setting. Whenever the AGC is currently being edited, the output headroom LED display changes to read gain-reduction. The makeup gain may be set manually by pressing the RATIO switch until the display reads **gAIn**. Set the makeup gain using the Wheel.
ATTACK Switch	Modifies the attack time of the AGC/Leveler (peak duration required to respond to a peak).
RELEASE Switch	Modifies the release time constant of the AGC. Remember that during no-signal periods, the AGC causes the release time to be infinite (gain-reduction release only occurs when there is no valid signal).
RATIO Switch	Adjusts the compression ratio of the AGC/Leveler between 1:1 (out) and 4:1. The compression ratio is the ratio of dB input change to dB output change. The compressor's makup gain may be set manually by pressing the RATIO switch until the display reads **gAIn**. Set the makeup gain using the Wheel.
THRESHOLD Switch	Sets the auto-release threshold (edit buffer offset 61). This is the level that a valid input signal must exceed to cause the AGC to readjust its gain to the new input signal. To adjust the actual compressor threshold (the compressor controlled by the auto-release software), use the realtime editor (**SEt** procedure) as described in Section 7.3.12 and edit parameter #56(dec).

4.5.6 Dynamics Section Control Summary

The following tables show the parameters used by each different section of the Dynamics Section. The first line of each table shows the function, the second line shows the front panel designation on the 602, the third shows the parameter name, and the fourth shows the parameter's range.

NR

THRESHOLD	ATTACK	RELEASE	FREQ/RATIO
threshold	n/a	n/a	frequency
-35 - 0 dB			1.0 - 21.11 kHz

DS (DE-ESS)

THRESHOLD	ATTACK	RELEASE	FREQ/RATIO
threshold	attack	release	frequency
-35 - 0 dB	0.1 - 9999 ms	100 - 9999 ms	31-21.11 kHz

EXPAND

THRESHOLD	ATTACK	RELEASE	FREQ/RATIO
threshold	attack	release	expansion ratio
-100 - 0 dB	0.1 - 9999 ms	100 - 9999 ms	1:1 - 1:8

COMPRESS

THRESHOLD	ATTACK	RELEASE	FREQ/RATIO
threshold	attack	release	compression ratio makeup gain
-100 - 0 dB	0.1 - 9999 ms	100 - 9999 ms	1:1 - 10:1 auto - 24 dB

AGC

THRESHOLD	ATTACK	RELEASE	FREQ/RATIO
auto release threshold	attack	release	compression ratio makeup gain
-100 - 0 dB	0.1 - 9999 ms	100 - 9999 ms	1:1 - 4:1 auto - 24 dB

4.5.7 Additional Dynamics Parameters

In addition to the previously mentioned controls, there are several additional parameters affecting the dynamics processor (Dynamic filtering, compressor and AGC). None of these controls are accessible directly from the front panel, however they may be accessed via MIDI or by means of the realtime editor (refer to Appendix A). If you modify any of these parameters and get lost, you can return to some semblance of reality by either loading program 256 and starting over or by loading any other program. Many of the preset programs modify these parameters also; this means that you can't simply duplicate a program by re-entering its front panel parameters.

These parameters are:

Name	Purpose	EditBuffer Offset
Dynamics sidechain filter mode	Changes the sidechain filter from highpass shelving to lowpass.	0
Expander knee control	Sets number of dB required to reach ultimate expansion ratio.	49
Compressor knee control	Sets number of dB required to reach ultimate ratio.	55
AGC Threshold	Absolute threshold for AGC	56
AGC knee control	Sets number of dB required to reach ultimate ratio.	60
ARM peak release time constant	Determines the recovery time of the auto-release monitor system.	62
ARM integration time constant	Affects the signal level history of the auto-release monitor system. Shorter times require higher peak-to-average ratios to release the AGC hold.	63
ARM Signal/Noise threshold	Sets peak/average ratio for signal/noise decision	64
Dynamics control chain turnover frequency	Shelving highpass filter to limit subsonic response of control chain (sidechain).	65
Log converter time constant	Initial log averaging time constant for the dynamics sidechain.	66
Lookahead delay time	Provides “thinking time” for dynamics sidechain, which can prevent overshoot.	67
De-ess absolute threshold	Sets threshold for onset of de-ess action. The relative threshold affects the degree of de-ess action after exceeding the absolute threshold.	22

4.5.7.1 Sidechain filter

The dynamics section sidechain has a shelving highpass filter in its control chain that limits the response of the dynamics section to very low-frequency sounds. The frequency (edit buffer 65) and mode (shelving highpass or lowpass, edit buffer 0) of this filter may be varied. The default condition is shelving highpass.

In general, raising the filter frequncy, in highpass mode), makes the dynamics section (compressor, AGC, and downward expander) less responsive to low-frequency sounds. This may be useful for preventing p-pops from causing compression or opening the downard expander.

Lowering the filter frequency, in lowpass mode) makes the dynamics section less responsive to high-frequency sounds. This may be useful for preventing sibilance or high-frequency noise (hiss, clicks, etc.) from triggering the dynamics section.

4.5.7.2 Expander knee control

The point in the downward expander's gain curve immediately below threshold is known as the knee. The width of the knee may be altered to make the transition to the expander's ultimate slope more or less gradual. Edit buffer 49 controls the downward expander's knee width.

4.5.7.3 Compressor knee control

The point in the compressor's gain curve immediately above threshold is known as the knee. The width of the knee may be altered to make the transition to the compressor's ultimate slope more or less gradual. Edit buffer 55 controls the compressor knee width.

4.5.7.4 AGC absolute threshold

The AGC normally acts as a compressor having its threshold level set *very* low. This parameter (edit buffer 56) controls just how low the actual threshold is. Below this level, there is no AGC action.

4.5.7.5 AGC knee control

The point in the AGC's gain curve immediately above threshold is known as the knee. The width of the knee may be altered to make the transition to the AGC's ultimate slope more or less gradual. Edit buffer 60 controls the AGC knee width.

4.5.7.6 ARM peak release TC

This parameter (edit buffer 62) affects the recovery time of the auto-release monitor (ARM) subsystem. Normal settings are in the 1 to 3 second range and the default setting is 2.5 seconds. Refer also to Section 4.5.7.7.

4.5.7.7 ARM integration TC

This parameter (edit buffer 63) affects the signal level history of the ARM subsystem. Shorter time constants require higher signal peak-to-average ratios to trigger the AGC hold function (thereby releasing the gain reduction). This time constant and the time constant used for offset 62 should be in the same range of 1 to 3 seconds.

4.5.7.8 ARM Signal/Noise threshold

The AGC uses a signal/noise detector (ARM or Auto-Release Monitor) to decide when to allow the AGC compressor's gain reduction to recover to it's no-signal value. The detector uses the peak-to-average ratio of the signal to decide whether the signal is noise or not noise. Raising the ARM signal/noise threshold (edit buffer 64) causes the detector to reject signals lacking much peak content. Lowering the threshold makes the detector less picky, eventually allowing noise to pass as signal.

4.5.7.9 Log converter time constant

The dynamics section's log converter converts the audio signal into a logarithmic representation of its signal level. The log converter time constant is a simple time constant at the output of the log converter. The time constant sets a minimum attack and release time for any signal. Some smoothing is necessary to prevent the compressor (or other dynamics processor) from trying to follow the envelope of low-frequency signals.; Edit buffer 66 controls the time constant.

4.5.7.10 Lookahead delay time

Overshoot is a problem with any compressor that is caused by the control signal arriving at the gain-controlled element (VCA in analog units) after the leading edge of the audio signal. A simple remedy for overshoot is to slightly delay the audio before it gets to the gain-controlled

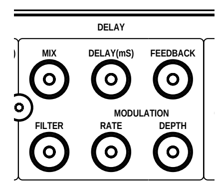
element; in essence giving the compressor “time to think”. Edit buffer 67 controls the lookahead delay time.

4.5.7.11 De-ess absolute threshold

The relative de-esser threshold can be accessed via the front panel. This threshold setting must be relative in order that it not be sensitive to the overall signal level at any given instant in time. The de-esser determines the relative amount of sibilant energy in the input signal which is then compared to the low-frequency content of the input signal. The high-frequency content must exceed the absolute threshold level. Once the signal exceeds the absolute threshold, the sibilant energy content must then exceed the relative threshold setting. Only then will de-essing occur. Edit buffer 22 controls the absolute threshold. Edit buffer 23 controls the relative threshold and may be adjusted via the front panel, the realtime editor, or via MIDI.

4.6 Delay Group

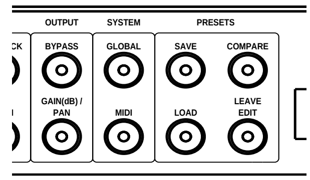
The delay group encompasses a dual-delay line with cross-coupled feedback and delay-time modulation. The delay time of each delay can be set independently, and the duAL mode changes the delay times of both delays simultaneously while maintaining any difference in the delay times. You can also add modulation to the delay time(s). The modulation signal may be random, a sine wave, or a triangle wave. The modulation signal depth (amplitude) and rate (frequency) are adjustable. The delay group creates effects ranging from simple slapback through small-room simulation, chorusing and flanging.



MIX Switch	Sets the MIX ratio (%) between the direct and the lowpass-filtered delayed signal. Holding down the Mix switch (**rt**) allows editing the rate of change of the mix gains. This parameter's rate-of-change is shared with the FEEDBACK switch.
DELAY Switch	**dL 1/dL 2/duAL** The two delay lines can be modified either individually or in tandem. The display indicates the delay time in milliseconds, and the DELAY switch toggles through the adjustment modes. Adjustment in tandem (**duAL**) changes the delay time of both delays simultaneously. This mode maintains any difference in delay time between the two delays. For monophonic delays, set both delay lines to the same delay time prior to entering **duAL** mode. In **duAL** mode, the display indicates the delay time of delay number one. Note: the delay-time increments used change depending upon the current delay time. At short increments, the change-increment is small (1 ms) growing to 2 ms above 30ms, growing to 5 ms above 100 ms. Above 310 milliseconds, the delay-time increment is 10 ms. Thus, as you traverse the range of delay settings, the increment between the two delays changes according to the increment used for that particular delay time. You can see the actual setting of each delay by toggling the DELAY switch through **dL 1** and **dL 2**. Refer also to the delay time table, found in Appendix C.

DELAY switch (cont'd)	A long press on the DELAY switch (**rt**) accesses the delay line rate of change, with 0.1 being basically instantaneous, and 9999 being very slow. This adjustment, along with the MODULATION DEPTH and RATE are used for chorus type effects or flanging.
FEEDBACK Switch	**n-90/oFF/P-90** Sets the attenuation of the cross-coupled feedback. The range of control is from off to 0 dB attenuation. The feedback signal may be in-phase (**P-nn**) or out-of-phase (**n-nn**) where nn corresponds to the amount of attenuation applied to the feedback signal. Thus P-10 corresponds to in-phase (positive) feedback, 10 dB down from unity gain. Both channels are adjusted simultaneously. The feedback polarity (phase) is especially important when creating flanging effects. Pressing this switch repeatedly toggles between '**out**' and the current feedback setting.
FILTER Switch	The signal from the delay line drives a single-pole (6 dB/octave) lowpass filter with a range of 600 to 18kHz. The output of the lowpass filter then feeds the feedback and mix controls.
RATE Switch	Sets the rate (frequency) for the delay-time modulation generators (DTMG). The DTMG is either a random number generator whose value is updated *rate* times per second, a sine wave of *rate* Hz, or a triangle wave of *rate* Hz, where *rate* is the value shown in the display. Pressing and holding the RATE switch allows changing the DTMG from random (**rAnd**), to a sine wave (**SinE**), or to a triangle wave (**AnGL**).
DEPTH Switch	Sets the depth of the modulation applied delay time of the variable delay lines. **100** is maximum, **0** is off. Pressing this switch toggles between '**out**' and the current modulation depth setting.

4.7 Output Group



The output group encompasses the switches affecting the output of the 602.

BYPASS Switch — The BYPASS switch puts the 602 into bypass mode. This is not a hard-wire bypass; the signal continues to flow through the A-D and D-A converters; the DSP processing is simply disabled.

GAIN/PAN Switch — The GAIN/PAN switch sets the output gain and left-right panning of the 602. A long press on the GAIN/PAN switch sets the rate-of-change of the level functions; the display indicates **rt**. The GAIN setting is saved on a per-program basis.

L /--.-- The display indicates the output gain setting (**L**) in dB or panning (**--.--**). The pan display indicates the percentage towards the left or right. Thus, **50.--** indicates 50% towards the left. The GAIN and PAN parameters are saved on a per-program basis. The PAN setting is saved on a per-program basis.

4.8 System Group

The system switches determine global (state saved with unit, not with programs) operating parameters, including MIDI. All global parameters are stored in battery backed-up memory; thus they are retained even in the absence of AC power.

4.8.1 Global Switch

The GLOBAL switch is a multi-mode switch with the following modes:

GAIn Sets digital input gain over a +/- 18 dB range before any digital processing modules. The gain setting is saved on a per-program basis.

InP Input selector. Rotating the Wheel selects the input source as indicated by the Digital/CH1/Stereo/CH2 LEDs. The input source may be the digital input (AES/EBU or S/PDIF), the line inputs, or the mix of the line inputs. The Digital In/Sync LED flashes if the digital input signal is missing or defective. Each input source may be routed to the inputs of the DSP as depicted by the display. In stereo mode, ‘1 . 2’, the gain controls for the two input channels may be split (one channel per control) or ganged (two channels on one control) by further rotating the Wheel after the LED display indicates. ‘1 . 2’.

Display Shows	Description
1.1	Input 1 routes to both outputs.
2.2	Input 2 routes to both outputs.
1.2	Input 1 and input 2 are mixed and routed to both outputs..
1 . 2	Stereo. The two inputs are separate. Further rotation of the wheel gangs the two gain controls onto the Channel 1 control and only the Ch1/Stereo LED indicator illuminates.

As you rotate the Wheel, the 602 cycles through the line and digital inputs. For each input, the 602 cycles through the four routing options shown in the table. For analog sources, inputs 1 and 2 are the left and right channels, respectively, of the digital-to-analog converter. For digital sources, inputs 1 and 2 are the left and right channels, respectively, of the digital input stream.

bAr1/bAr2 Temporarily shifts the INPUT HEADROOM display to monitor one or the other of the two input channels. The measurement point is immediately prior to the digital gain trim. Exiting this menu function restores the bargraph to its normal mode, which displays the highest peak level of the two input signals.

CLCI / CLCE Clock source for the internal ADC, DAC, and DSP. If the display reads **CLCI** and if the input source is microphone or line then the clock source is the internal 44.1 kHz or 48 kHz sample rate oscillator.

If the display reads **CLCE** then the external AES / EBU input is used for the clock reference. This allows a master system clock to provide the sample-rate reference for the 602, and precludes any problems with mismatch or drift between digital clocks.

nP-- / Prt Controls memory write protection. When memory is not write-protected the SAVE switch LED will either be on solid (edit buffer not dirty) or flashing (edit buffer modified / dirty). When memory protection is enabled, the SAVE switch LED never illuminates.

44.1 / 48.0 Selects 44.1kHz or 48kHz as the internal sampling rate for analog input signals. This menu selection is only valid for internal clock sources (**CLCI**).

4.8.2 MIDI Switch

The MIDI switch sets various MIDI parameters in the 602. In addition, it accesses the realtime block editor allowing modification of either of the two realtime blocks or setting any parameter within the edit buffer. A short press on the MIDI switch accesses the following parameters:

CH.nn Sets the MIDI channel number where **nn** can be **AL** for omni mode or 1-16 for a specific MIDI channel number. Stored as global parameter.

U.nnn Sets the MIDI unit number. Allows multiple 601s to share the same MIDI channel number for sysex type messages. **nnn** can range from 0-7e for specific unit numbers, or **AL** to ignore the unit number in sysex messages. Stored as global parameter.

dnEd Downloads the edit buffer. Holding down the switch (long press) sends out the current state of the edit buffer as a sysex message. After the complete buffer has been sent the display reads **donE**.

dnAl Downloads all stored programs and all ROM programs. Holding down the switch (long press) sends out all programs. After the dump is complete the display reads **donE**. During the dump the decimal point walks to show progress.

rEAL Allows creating realtime MIDI setups as well as setting any edit buffer parameter from the front panel. For a complete discussion, refer to Chapter 7 and Appendix A of this manual.

A long press on the MIDI switch accesses the realtime block editor. This is described in greater detail in Appendix A of this manual.

4.9 Presets Group

This group of switches handles memory and program related tasks.

SAVE Switch

The SAVE switch saves the contents of the edit buffer to the selected memory location. The SAVE switch flashes if the edit buffer has been modified (is dirty) and memory protection has not been enabled. If the edit buffer is clean and memory protection disabled, the LED illuminates steadily. The LED is off when program memory is protected. When memory protection is on, trying to save the edit buffer displays the **Prt** (protected) message. With memory protection off, a long press saves the edit buffer using the currently displayed program number (number seen when not in edit mode). The display shows **donE** after the save operation has completed. Remember that program numbers above 128 are reserved and always write protected.

The following table shows the effects of memory protection, the edit buffer state, and the SAVE switch.

Edit Buffer	Memory Protection	SAVE LED	SAVE switch	Display reads
clean	disabled/off	steady	no save	**- -**
clean	enabled/on	out	no save	**Prt**
dirty	enabled/on	out	no save	**Prt**
dirty	disabled/off	flashes	program saved	**donE**

COMPARE Switch

The COMPARE switch toggles the 602's settings between those stored in the edit buffer and those stored in program memory. This allows making quick a/b comparisons between the original program and the current settings. The display toggles between OLD and CURR to help you keep things straight.

LOAD Switch

The LOAD switch loads a copy of the program whose number currently shows in the display into the edit buffer for editing. The display reads **donE** when the operation is complete.

LOAD Switch (cont'd)

If the program number has been changed with the Wheel, the LOAD switch LED and the preset number shown in the display flashes, indicating that a new program is available for loading. If the program number has not been changed, but you still want to load the original program over the current edit buffer (i.e. start over), holding down the LOAD button forces a load operation. In either case, the display shows DONE when complete.

LEAVE EDIT Switch

The LEAVE EDIT switch terminates any editing operation without disturbing or destroying the contents of the edit buffer. You use this switch to return to the top-most control mode (program number shows in display).

4.10 Setting Scenarios

The following scenarios may help clarify setting up the 602 for various analog and/or digital input signals.

Situation 1: Internal ADC, internal sample clock, DACs fed from DSP.

- ❒ Under globals, click to **InP**. Set left-hand input LEDs to Ch1/Stereo and/or Ch2.
- ❒ Click globals again to **C--- or CLCI** or **CLCE**. Set to **CLCI** for internal master sample clock.
- ❒ Click globals again to **44.1** or **48.0.** Select sample rate.

Situation 2: Internal ADC, external sample clock, DACs fed from DSP.

- ❒ Under globals, click to **InP**. Set left-hand input LEDs to Ch1/Stereo and/or Ch2.
- ❒ Click globals again to **C--- or CLCI** or **CLCE**. Set to **CLCE** for master sample clock from AES/EBU reference.
- ❒ Connect external sample clock source to AES/EBU input.

Situation 3: Input from AES/EBU, external sample clock, DACs from DSP.

- ❒ Under globals, click to **InP**. Set left input LEDs to digital.
- ❒ Click globals again to **C---** or **CLCI** or **CLCE**. Clock and signal source will be forced to external AES/EBU.

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Note: The next two sections are presented on their own page. This makes it easy to remove the page should you want to prevent other readers from knowing how to initialize the 602 or bypass the front-panel security features.

4.11 Restoring Factory Presets

Caution

Do not reinitialize the 602 *to the factory set values unless this is what you* ***really*** *want to do. Reinitializing* ***erases all*** *user programs (presets 1 to 128) and there is no way to recover your programs once you have done this.*

You can reinitialize all programs to their factory set values by holding down the **load** switch while applying power to the 602.

4.12 Disabling the Front Panel

Some applications may require disabling the front panel. Broadcasters using the 602 as their on-the-air mic processor may want to make the unit impervious to adjustment.

The 602 has three levels of security:

None This is what you normally get when you turn the 602 on.

Partial Disables everything except the Wheel and the **load** button. Since no other buttons operate, it is impossible to alter programs or to overwrite other programs.

Maximum Disables everything. **Nothing** on the front panel works.

To activate (or deactivate) any of the security features, turn the 602 off, press and hold one of the following buttons, and turn the 602 on.

FILTER Enables no security. Everything is accessible.

RATE Enables partial security.

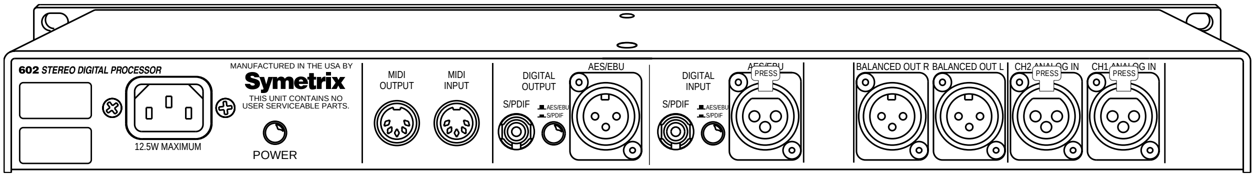
DEPTH Enables maximum security.

With partial or maximum security enabled, attempting to access any secured function results in a LoC indication on the display. You can also defeat any security feature in effect by reinitializing the 602.
Reinitializing erases ALL user programs (presets 1-128).

The security features can be activated via MIDI. All 602 functions are always MIDI-accessible regardless of the security level.

Notes

5. Rear Panel Overview



Serial Number	Do yourself a favor and write this number down somewhere safe, and while you're at it, please send us the completed warranty card.
AC Power Input	IEC-power connector. Connect only to appropriate AC power source. Refer to actual rear-panel marking for correct AC source value.
POWER switch	Push-push switch turns the 602 on and off.
MIDI connectors	5-pin DIN connectors used for MIDI output and input.
DIGITAL OUTPUT	RCA connector and XLR-male connector used for S/PDIF and AES/EBU (respectively) digital output. Push-push switch selects between protocols.
DIGITAL INPUT	RCA connector and XLR-female connector used for S/PDIF and AES/EBU (respectively) digital input. Push-push switch selects between protocols. All connectors transformer isolated.
Outputs	XLR-male, balanced. Analog audio output of the 602. Pin 2 is hot.
Inputs	XLR-female, balanced, line level analog inputs. Pin 2 is hot.

Notes

6. Fast First Time Setup

Follow these instructions to get your 602 up-and-running as quickly as possible. The intent of this section is to get the 602 to pass signal. If you need something clarified, you'll find the answer elsewhere in this manual.

Figure 6-1 is a simplified block diagram of the 602. Take a moment now, check the block diagram out, and take note of the following points:

- ❏ The diagram shows three different signals: mono, stereo, and data.
- ❏ All input signals are treated as a stereo pair; any processing applied applies equally to both channels.
- ❏ Line level signals are converted to digital and applied equally to the left and right digital inputs of the DSP chain.
- ❏ Stereo signals applied via the digital inputs remain stereo.

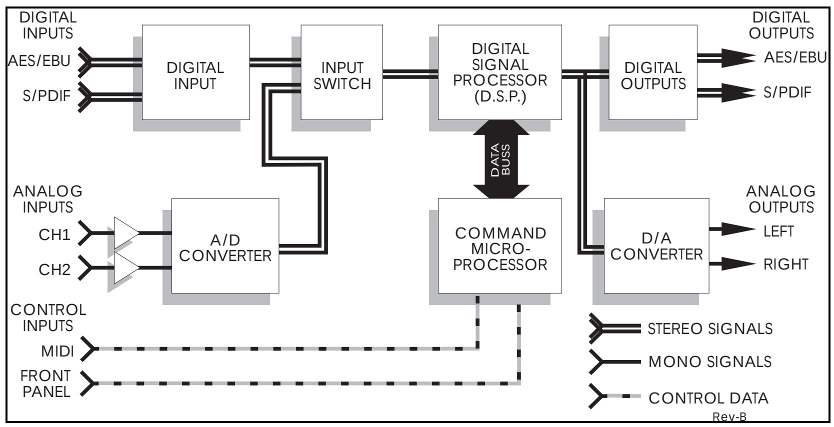


Figure 6-1. Simplified block diagram

- ❏ Mono signals applied via the analog inputs can emerge from one or both outputs (if you use the delay with some difference in the delay times, you can "stereoize" the output).

6.1 Connections

Connect your analog input source to the appropriate XLR connector. The line inputs are intended for balanced or unbalanced line level inputs with signal levels between -10 and +4 dBu. Connect the 602's analog outputs to your console's line inputs using the XLR, connectors.

The digital input/output connectors are intended for sources or loads conforming to the S/PDIF or AES/EBU digital interface standards. The 602's digital input accepts any word length up to 24-bits.

The analog and digital inputs and outputs may be used in any combination (i.e. analog in - analog out, analog in - digital out, digital in - digital out, digital in - analog out.) The 602 operates at either 44.1 kHz or 48.0 kHz sample rates (input and output are always the same rate).

If you are using the 602's analog inputs with the digital outputs, you can supply an external sample-rate reference signal via the S/PDIF or AES/EBU digital input. This may be useful in situations using a single master clock source. Designate the digital input as the clock source via the GLOBAL parameter switch, parameter **CLCE**.

If you are using the analog outputs, connect them to your console's balanced line inputs. If you are driving an unbalanced input, pin 3 of the XLR connector should float. If your audio system uses pin 3 of the XLR connector as the "hot" connection, then pin 2 of the XLR connector **must** float. This is described in greater detail in Chapter 3.

If you are using the digital inputs, connect them to an appropriate digital source. Set the push-push switch to correspond to the input that you are using.

If you are using the digital outputs, connect them to an appropriate digital input. Set the push-push switch to correspond to the output that you are using.

There is no need to observe polarity with regard to either of the AES/EBU I/O connectors. The digital system is immune to polarity reversals on the signal wiring.

Connect the AC input to an AC power source of the proper voltage and frequency, as marked on the rear of the unit.

Caution: *Failure to connect the* 602 *to the proper AC mains voltage may cause fire and/or internal damage. There are no user serviceable parts inside the chassis. Refer all service to qualified service personnel or to the factory.*

Warning: Lethal voltages are present inside the chassis. There are no user serviceable parts inside the chassis. Refer all service to qualified service personnel or to the factory.

6.2 Settings for Analog Sources

For an analog source, Figure 6-2 and Figure 6-3 show the wiring required. Set the controls and switches on the front panel as follows:

1. After all rear-panel input and output connections have been made, apply power to the 602 and depress the Power switch. When the display shows program number 1, proceed to the next step.
2. Depress the Global switch once. The display reads GAIn. Rotate the Wheel to set the digital input gain (this is not the gain applied to AES/EBU or S/PDIF sources) to 0.
3. Depress the Global switch again. The display reads InP. Rotating the Wheel selects the input source and routing as indicated by the Digital/CH1/Stereo/CH2 LEDs and the display. The input possibilities are: digital or analog line input. Rotate the Wheel to select the digital or analog source, and your desired routing. For analog stereo operation, rotate the Wheel until the CH1/STEREO and CH2 LEDs illuminate and the display indicates ‘1 . 2’
4. Depress the Global switch again; the display reads bAr1.
5. Depress the Global switch again. The display reads CLCI. Rotating the Wheel selects the clock source for the digital processors. If set to CLCI the clock source is the internal 48 kHz or 44.1 kHz crystal oscillator. If the display reads CLCE the clock source is the rear-panel digital input. Refer to Chapter 4 for additional information. Select **CLCI.**
6. Depress the Global switch again. The display reads nP-- (not protected) or **Prt** (protected). Select memory protection as required.
7. Depress the Global switch again. The display reads 44.1 or 48.0. This represents the two sample rates (only if you haven’t selected CLCE in step 5. Rotate the Wheel to select the sample rate appropriate to your application.
8. Depress the level switch. The display reads L 0.0. If not, rotate the Wheel until the display reads L 0.0. This sets the output gain to 0 dB.
9. Set the input level by increasing the setting of the selected input level control until the green LEDs in the HEADROOM display illuminate. Ideally, the highest signal level should illuminate the -2 dB LED, and the CLIPPING LED should never illuminate (the CLIPPING LED operates at 1 dB below clipping.)

10. It is possible for the Clipping LED to illuminate even though the green LEDs in the Headroom display are not completely illuminated. If this occurs, decrease the setting of the appropriate gain trim control sufficiently to keep the Clipping LED from illuminating, then access the global digital **gAIn** setting from the GLOBAL switch. Increase the digital **gAIn** setting as required.

11. The 602 should now pass signal.

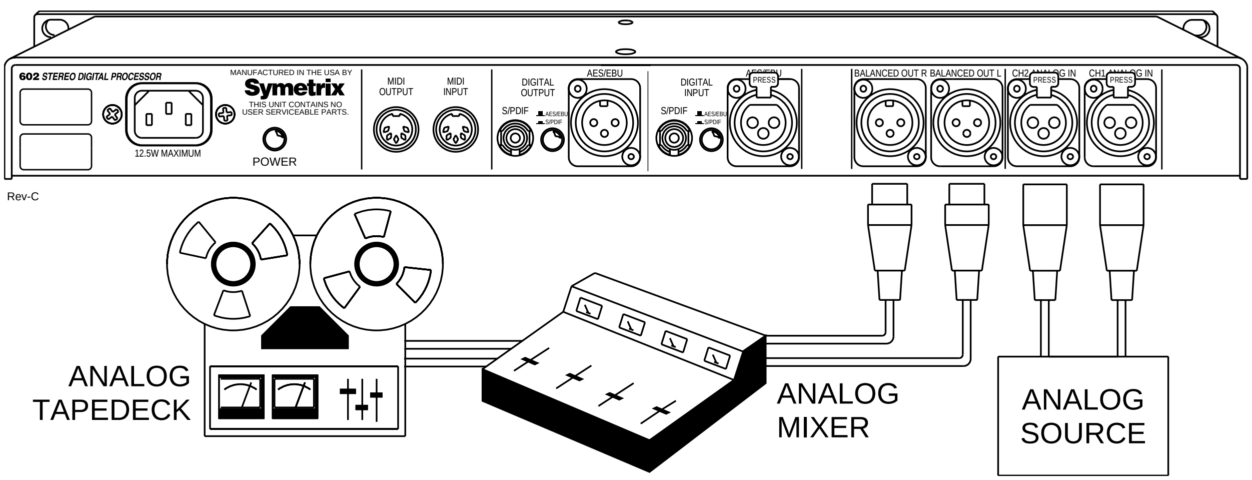


Figure 6-2. Using the 602 to process a source.

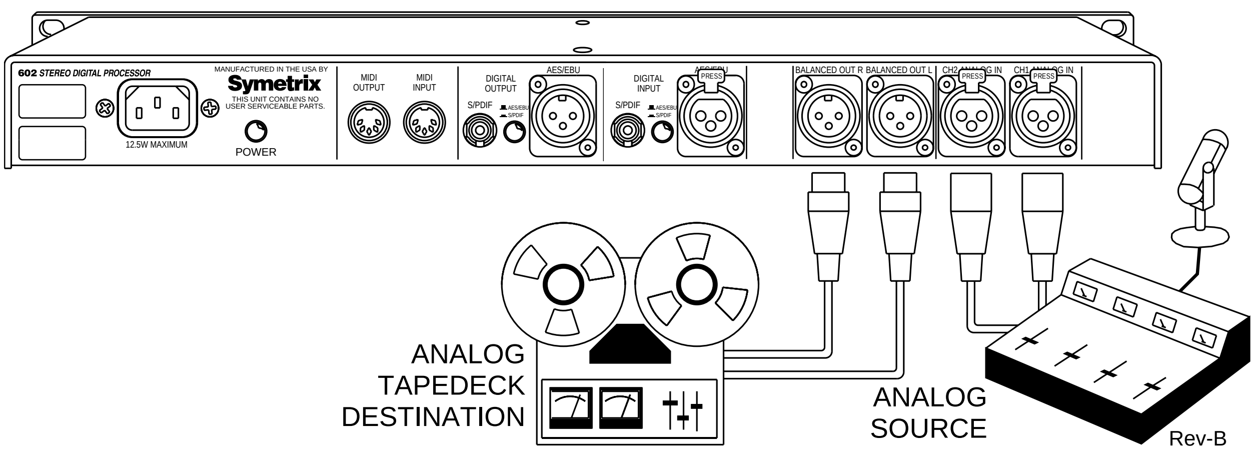


Figure 6-3. Using the 602 to process an entire mix.

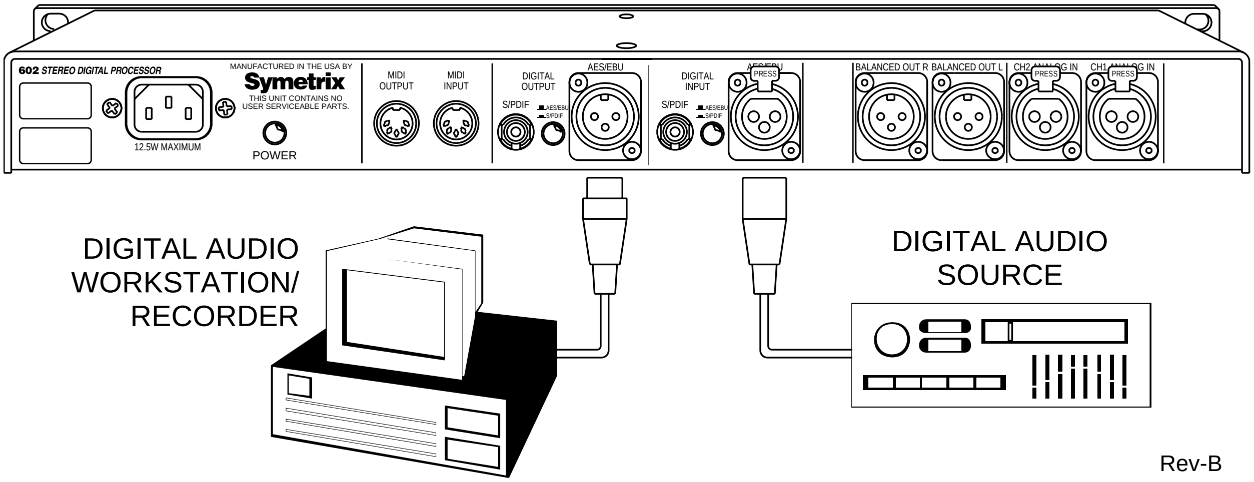


Figure 6-4. Using the 602 with a digital source and digital destination.

6.3 Settings for Digital Sources

For a digital source, Figure 6-4 shows the connections required. Set the controls and switches on the front panel as follows:

1. After all rear-panel input and output connections have been made, apply power to the 602 and depress the Power switch. When the display shows program number 1, proceed to the next step.
2. Depress the Global switch once. The display reads GAIn. Rotate the Wheel to set the digital input gain to 0.
3. Depress the Global switch again. The display reads InP. Rotating the Wheel selects the input source and routing as indicated by the DIGITAL/CH1/STEREO/CH2 LEDs and the display. The input possibilities are: digital or analog line input. Rotate the Wheel to select the digital or analog source, and your desired routing. For analog stereo operation, rotate the Wheel until the CH1/STEREO and CH2 LEDs illuminate and the display indicates ‘1 . 2’
4. Depress the Global switch again. The display reads bAr1.
5. Depress the Global switch again. The display reads CL--. The clock/sample-rate reference is the external digital signal. Refer to Chapter 4 for additional information.
6. Depress the Global switch again. The display reads nP-- (not protected) or Prt (protected). Select memory protection as required.
7. Depress the level switch. The display reads L 0.0. If not, rotate the Wheel until the display reads L 0.0. This sets the output gain to 0 dB.

Set the input level by accessing the global **gAIn** setting from the GLOBAL switch. This parameter is the digital input gain. Increase the **gAIn** setting as required so that the -2 dB LED in the HEADROOM display illuminates. Since the red CLIPPING LED is driven from the analog inputs, it should never illuminate. The 602 should now pass signal.

7. Using the 602.

This chapter is intended for more advanced users. If you are a first-time user, we recommend that you start out by using the procedure found in "Fast, First-Time Setup." Elsewhere in this chapter, you can find operational hints and suggested settings. You can find additional discussion of many of these topics in Chapter 2, "Basics."

7.1 Installation

The 602 may be installed free-standing or rack mounted. No special ventilation requirements are necessary.

Installation Requirements	
Mechanical	One rack space (1.75 inches) required, 12.5 inches depth (including connector allowance). Rear chassis support recommended for road applications.
Electrical	105-125 V ac, 60 Hz, 20 watts. 210-250V ac, 50 Hz, 20 watts (export version).
Connectors	XLR-3 female for inputs, XLR-3 male for outputs, Pin 2 of the XLR connectors is "Hot." RCA female connectors for S/PDIF digital I/O. XLR-3 male and female connectors for AES/EBU digital I/O.

7.2 Operational Details

This section describes the details of operating the 602. Usage information can be found later in this chapter.

The 602 accepts stereo or mono analog, line-level, input signals, converts them to 18-bit digital form, splits them into left and right signals, processes them through two parallel DSP chains, and then converts the signals back to the analog domain. The processed signals are also simultaneously available at the AES/EBU or S/PDIF output connectors.

The 602 can also process the microphone input through one channel and the line input through the other channel. The control signal processing is still shared between the two channels.

Digital signals at either 44.1 kHz or 48 kHz sample rates may be fed directly into the 602 for processing. The processed signals are available at the outputs as AES/EBU or S/PDIF, and stereo analog balanced line level. The digital outputs and analog outputs operate simultaneously. The 602 does not perform dithering or re-dithering.

Regardless of the input source, the 602 always treats its input signals as a stereo pair. With the exception of the delay line, the 602 always applies identical processing to both signals. Digital signals may be up to 24-bits wide; the 602 treats all digital signals as if they were 24-bit.

The equalizer is a digital implementation of a common three-band parametric equalizer. The usual complement of controls may be found and the outside bands may be converted into shelving equalizers. All bands cover the entire frequency range.

The DS and Noise Reduction block have independent control chains. The Noise Reduction system implements a variable-frequency low-pass filter controlled by on the relative high-frequency content of the input signal. The De-esser is a broadband limiter having a sharply peaked filter in its sidechain.

The Dynamics block is a digital realization of an analog compressor/AGC/expander. A common log converter provides a logarithmic representation of the amplitude of the input signal to the components of the Dynamics block. Within the dynamics block, the component having the greatest amount of instantaneous gain reduction controls the gain of the digitally controlled attenuators (DCA). An adjustable delay before the DCA allows controlling the amount of overshoot occurring within the compressor.

The Delay block uses two delay lines (one per channel) with their recirculation paths cross-coupled. The feedback signal may be polarity-inverted and the delay time may be controlled by an internal modulation oscillator or the front panel.

Many of the 602’s parameters have a rate-of-change parameter associated with them. This parameter determines how quickly the 602 responds to a step-change in the value of the parameter. This parameter ranges from 100 microseconds to 10 seconds. A 10 second rate-of-change setting makes the 602 change from the old value to the new value over a period of 10 seconds.

7.2.1 Stand-alone Operation

A vast majority of users use the 602 as a stand-alone device. Here the 602 replaces their usual complement of signal processing and either feeds their tape machine or workstation directly, in essence becoming a one-input, one-output console. If you are using the digital outputs of the 602, be sure to read sections 7.3.4. and 7.3.5.

If you are using the analog outputs, ensure that they are plugged into a line-level input (+4 dBu nominal level). If you have to plug the 602 into a microphone input (-40 dBu nominal level), then you'll need to pad (attenuate) the output of the 602 down to microphone level. A simple U pad is sufficient. A suitable design can be found in Figure 7-1. Although a far preferable connection would be to bypass your console or mixer's mic preamp, this will work. Ensure that there is no phantom power present at the console's mic input terminals (both sides of the mic input should read 0V dc referenced to ground).

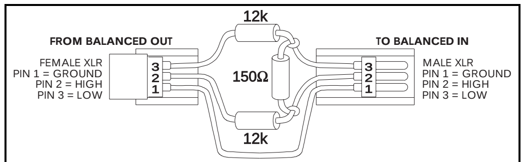


Figure 7-1. A 44 dB U pad.

Note: Padding (attenuating) the output of the 602 back to microphone level is a workable solution towards interfacing the 602 into a console or system having only microphone level inputs. However workable, the ultimate performance of the 602 will be limited by the performance of your system's existing microphone preamps. If you can find a way to bypass the existing microphone preamps in your system, do so. It'll be worth the trouble.

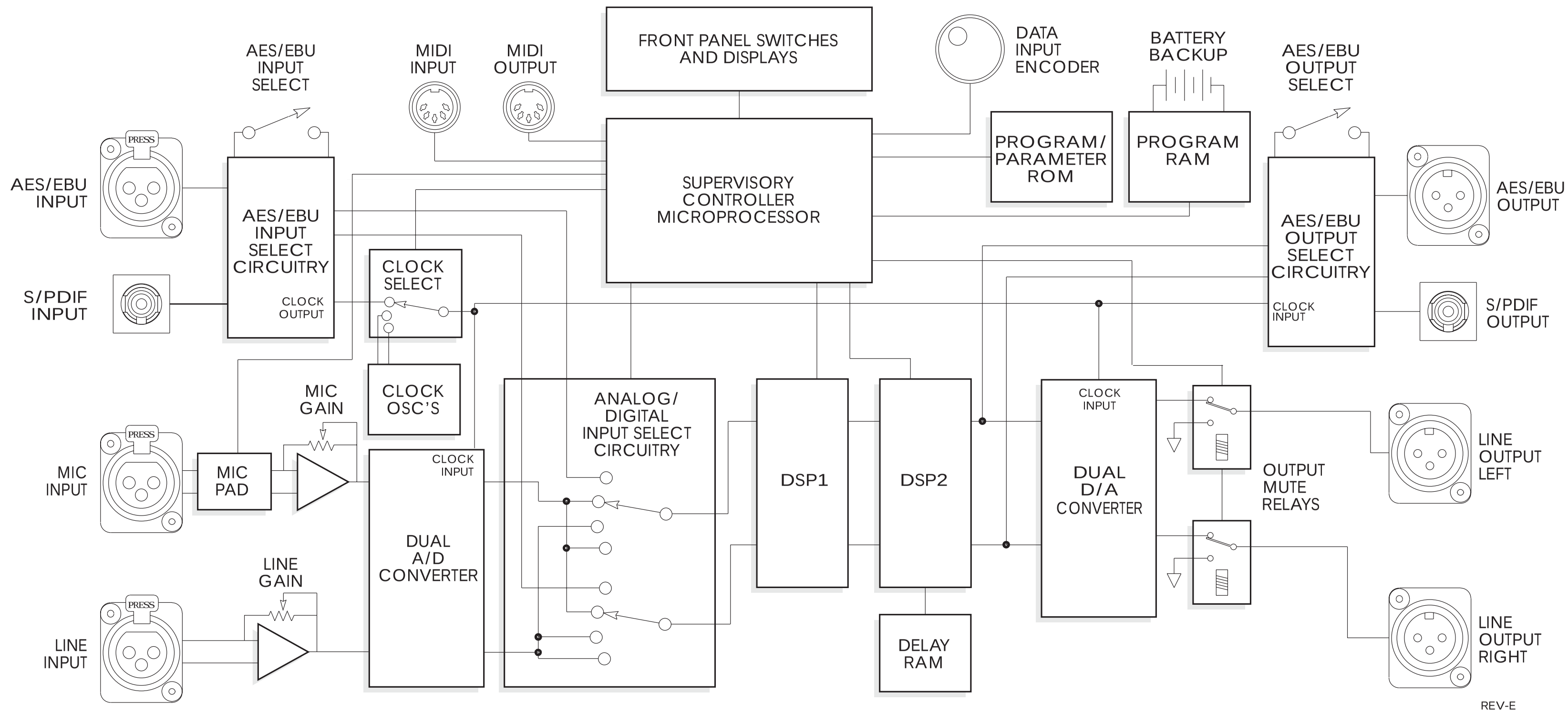


Figure 7-2. Overall block diagram.

7.3 Block Diagrams

On the preceding and following pages, you can find the block diagrams for the de-esser, dynamic noise reduction, dynamics processors, delay processor, and the entire 601. Please take a moment and take note of the following:

7.3.1 Overall Block Diagram

Refer to Figure 7-2.

- ❒ Two DSP chips handle all of the signal processing functions.
- ❒ The AES/EBU or S/PDIF inputs and outputs may be re-configured to connect between the DSP section and the D/A converter.
- ❒ The external digital inputs may also be used for an external clock reference.
- ❒ Presets and global parameters are stored in battery backed-up RAM.
- ❒ Both audio channels are always processed together. It is not possible to separate the two channels.

7.3.2 Sequence of Processing

- ❒ Note the order of the different signal processors.

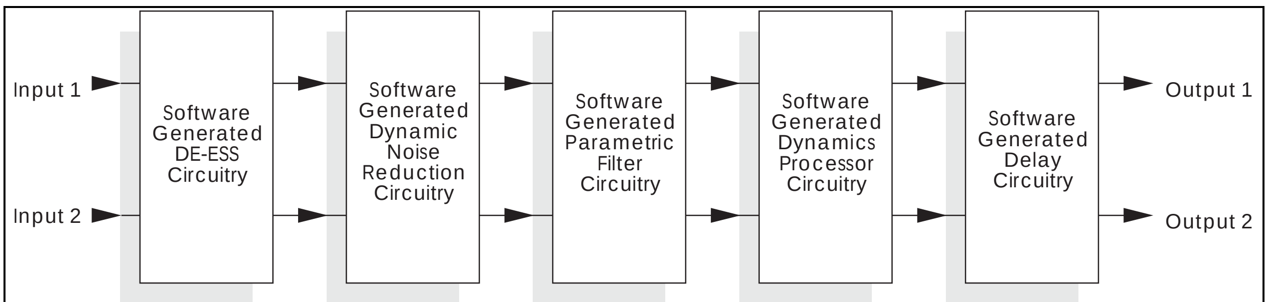


Figure 7-3. Sequence of Processing.

7.3.3 Dynamics Block

- ❑ The compressor and expander operate simultaneously. The gain reduction value is determined by the processor having the greatest gain-reduction output.
- ❑ The auto release circuitry operates when the AGC/Leveler is engaged.
- ❑ The signal path delay compensates for the computational time needed to compute the gain-reduction amount. For extremely short attack times, you may need to increase this parameter to allow the compressor to anticipate the input signal.
- ❑ Both channels always receive the same gain-reduction signal and the larger of the two input signals at any given instant becomes the source for any gain-reduction computations.
- ❑ The MIDI edit buffer parameter numbers are shown in parenthesis.

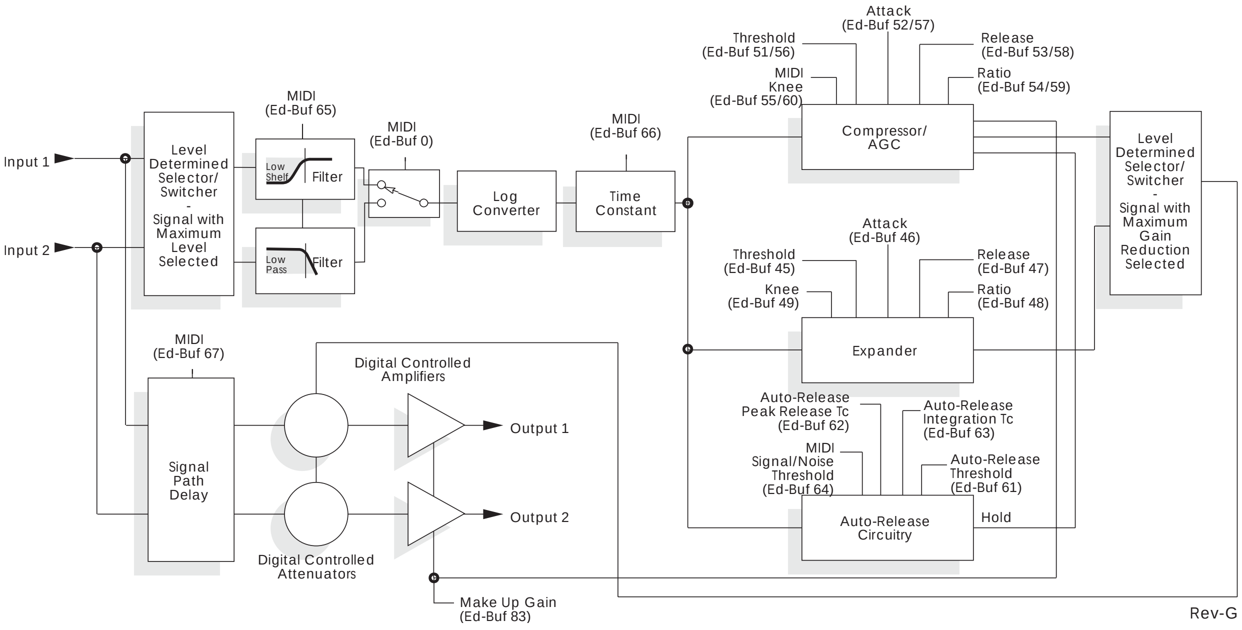
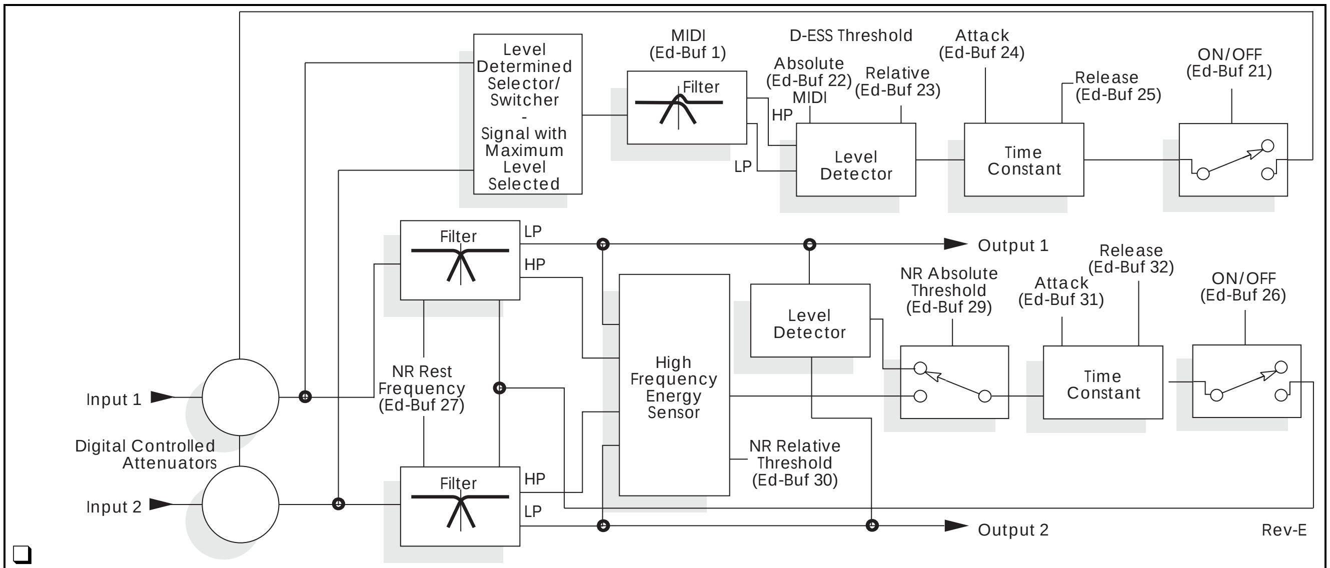


Figure 7-4. Dynamics block.

7.3.4 De-Ess and Noise Reduction Block

- ❑ The NR system uses a common control chain for both channels.
- ❑ The De-ess system uses a common control chain for both channels
- ❑ The MIDI edit buffer parameter numbers are shown in parenthesis.



❑

❑ **Figure 7-5. Noise Reduction and De-Ess block.**

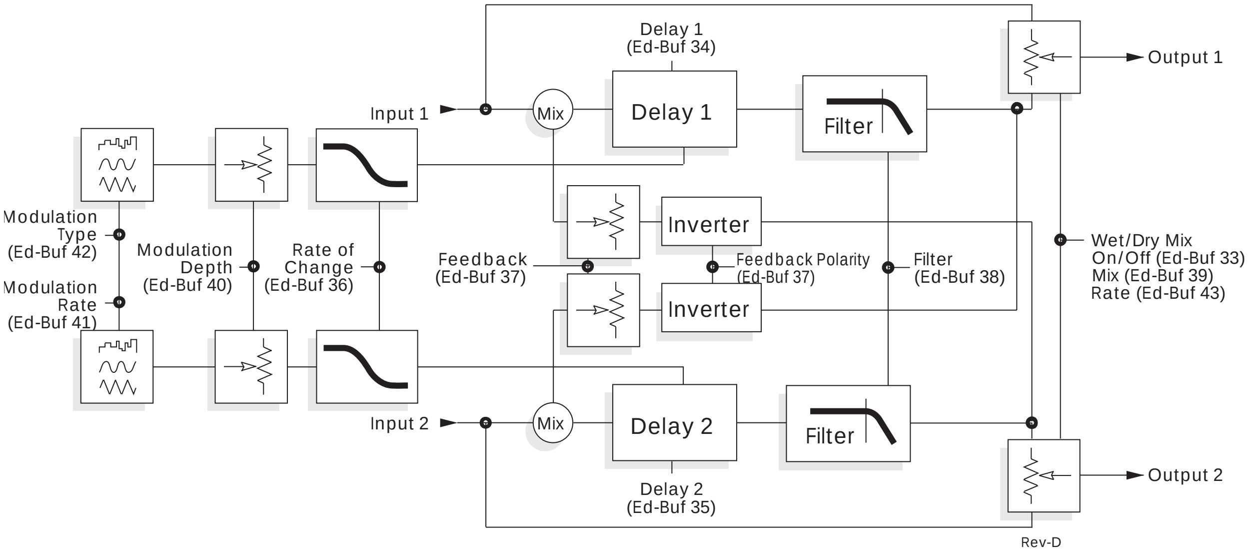


Figure 7-6. Delay block

7.3.5 Delay Block

- ❑ There are two delay lines, each independently adjustable.
- ❑ The feedback (recirculation) signals for the delay lines are cross-coupled.
- ❑ The feedback setting is always the same for both delays.
- ❑ For clarity, the diagram shows the "mix pots" reversed. When the mix parameter is 0, both outputs are their respective input signals. When the mix parameter is maximum, both outputs are the lowpass filtered delay line outputs.
- ❑ The MIDI edit buffer parameter numbers are shown in parenthesis.

7.4 System Interface

The 602 can be used in a variety of ways, some of which may be obvious, some of which may not be so obvious. The next portion of this chapter describes some of the different ways to use the 602.

7.4.1 Using the 602 as a Channel Insert Device

The 602 can also be used as a channel-insert device with your console. Use one or both of the 602's line inputs and one or both of the 602's line outputs. If you use both line outputs, then you'll need a second channel at your console for the 602's second line output.

7.4.2 Using the 602 in a Send-Receive Loop

The 602 can also be used in a console's send-receive (effects) loop. Drive the 602's line input(s) from the console's effects send and feed one or both of the 602's line outputs to your consoles effects returns. Ensure that the 602's delay mix parameter is set at 100%. If you use both of the 602's line outputs for stereo, then you'll need a stereo effects return or a second mono effects return for the second line output.

7.4.3 Using the 602 as an A-D Converter

You can use the 602 as an analog-to-digital converter simply by using the analog input(s) and the AES/EBU or S/PDIF output. There are, however, several caveats:

1. The clock accuracy specification stated in the AES/EBU standard is quite stringent In applications requiring simultaneous digital sources (like a digital mixer or digital multitrack recorder), the sample-rate clocks for every source should be phase-locked (synchronized) to a common source. This is described in section 7.3.5.

2. The 602 has internal 44.1 kHz and 48 kHz clocks, which should be adequate for most applications.

7.4.4 External Sample-Rate Clock

The 602 can be synchronized to an external clock signal via the AES/EBU or S/PDIF digital inputs. From the front panel, select **CLCE** (in the global block), Apply the external clock signal to either of the digital inputs. Avoid paralleling more than 2 inputs as loading of the clock source becomes a problem. The sample-rate clock should be a dedicated AES/EBU signal source.

7.4.5 Input/Output/Clock Summary

The following table tabulates the various input/output/clock possibilities. See also section 4.8.1.

Input Source Setting	Input LED Glows	Digital IN/SYNC LED	Sample Rate	Clock Ref.	Clock Setting	Notes
Analog	Line	off	44.1 kHz or 48 kHz	internal	**CLCI**	
Analog	Line	steady	either	external	**CLCE**	1, 2, 3, 5
Analog	Line	steady	either	external	**CLCE**	2, 3, 4, 5
Digital	Digital IN/SYNC	steady	either	external	**CL--**	1, 2, 3, 5
Any		flashing	either	external	**CLCE**	3, 5

Notes (correspond to "Notes" column in previous table):

1. Connect the digital clock source to the digital input connector.
2. The source connected to the digital input supplies the sample-rate clock.
3. The DIGITAL IN/SYNC LED flashes if there is no digital source, or if the digital data is faulty.
4. Connect the digital I/O connectors to another digital I/O processor.
5. The DIGITAL IN/SYNC LED column shows the state of the DIGITAL IN/SYNC LED on the front panel.

7.4.6 MIDI Programming

The 602 is MIDI programmable. At one level, you can simply send MIDI program change messages to load pre-stored programs (yours or the factory presets). At another level, you can manipulate program parameters via MIDI, and at yet another level, you can modify program parameters in realtime, during operation.

The 602 responds to the following MIDI messages:

1. Program Change
2. Control Change
3. Sysex
4. Pitch Bend
5. Aftertouch

You can theoretically operate up to 127 602s on a single MIDI bus. Depending upon how they are programmed, you can access them individually, or as a group.

7.4.7 Accessing Parameters via MIDI

All front-panel parameters may be accessed via MIDI. In addition, all secondary (hidden) parameters may be accessed via MIDI. A list of all accessible parameters may be found in Appendix A. These parameters may be altered via a MIDI sysex message or by using the procedure found in Section 7.4.10.

7.4.8 Realtime MIDI

Many of the 602's MIDI controllable parameters lend themselves to realtime control using a MIDI continuous controller. Some of these parameters are:

1. Output level and pan
2. Filter frequency and level
3. Delay mix.

An example of Realtime MIDI may be found in Appendix B.

7.4.9 Program Storage

The 602 provides non-volatile storage for 128 user programs (Program numbers 1 through 128). Program numbers greater than 128 are factory presets and are always protected. You can edit any of the factory presets and store it in one of the user program numbers. You can dump the contents of the 602's program memory to the MIDI OUT connector on the rear panel. Conversely, you can also load the 602's program memory via MIDI.

7.4.10 Editing Parameters not Accessible from the Front Panel

The front panel realtime update editing function can also be used to set the edit buffer value for parameters that are normally inaccessible from the front panel. These parameters are:

Processor	Offset (dec)	Parameter name	Reference
Dynamics Processor	0	Sidechain Filter Mode	0: Hipass Shelving 127: Lowpass
De-ess Processor:	22	Absolute Threshold	See Attn100 Table
Expansion Parameters	49	Expander Knee	See Knee Table
Compression Parameters	55	Compressor Knee	See Knee Table
AGC Parameters	56	Absolute Threshold	See Attn100 Table
	60	AGC Curve Knee	See Knee Table
ARM Sense Parameters	62	ARM Peak Release Tc	See Tc Table
	63	ARM Integration Tc	See Tc Table
	64	ARM Threshold	See ARM Threshold Table
LOG Converter Parameters	65	Control Chain Hipass Freq	See Frequency Table
	66	Log Averaging Filter Tc	See Tc Table
	67	Sidechain Lookahead	0: 0µS, 127: 2.6 ms @ 48kHz See Sidechain Lookahead Time Table

Modifying some of these parameters incorrectly can result in improperly operating modules, but reloading the program will restore the original settings. To edit one of these parameters use the realtime editing mode.

1. Cycle through the MIDI switch until the display reads **rEAL**, then hold down the switch. The display reads **bLC** (Block).
2. Use the Wheel to select **SEt.** Press the MIDI switch again. The display reads **PAr**.
3. Select the parameter to edit using the Wheel. (See Appendix C, "*Realtime MIDI*.") The OUTPUT HEADROOM display displays the 0-127 scaled level of the parameter. Press the MIDI switch again. The display reads **oFt** (Offset).
4. Use the Wheel to enter the desired value.
5. Press LEAVE EDIT to exit the realtime editing mode.
6. When using the realtime modulation modes, the offset adjustment sets the new value within a resolution of 2 steps.

Shortcut:

1. From any control level other than MIDI, press and hold the MIDI button to access the realtime editor. The display shows **bLC** (Block) when you are successful.
2. Use steps 2 through 6, above, to set the desired parameter.

7.5 Tips and Techniques for Using the 602

Following are some tips and techniques for using the 602. You should consider any settings given as starting points for developing your own settings. More general discussions of these topics may also be found in Chapter 2 of this manual.

7.5.1 Recalling and Storing Settings

Recall any program by pressing the LEAVE EDIT button, then using the Wheel to select the new program, then pressing the LOAD button. The new program has loaded when the display reads **donE**. The 602 always loads a copy of the program into the edit buffer (regardless of whether you want to edit the program or not). The program in the edit buffer is also the program that the 602s processor executes, unless the COMPARE button has been pressed, in which case the 602 executes the program out of the program RAM/ROM.

When you modify the edit buffer by changing any parameter, the SAVE switch flashes unless the program memory has been write-protected (either by location or via the global parameter block). If the SAVE switch is flashing, the edit buffer is dirty; that is, its contents have changed. The modified program can be stored in program numbers 1 through 128. If you try to save a program to presets 129 through 256 (which are always protected) the display reads **Prt**, indicating that the selected program number is protected (read-only). Chose a program number between 1 through 128 for your program.

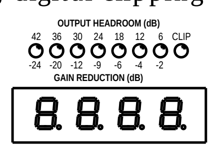
Store a modified program by pressing the LEAVE EDIT button, then using the Wheel to select a program number for the modified program (remember, between 1 and 128), then pressing and holding the SAVE button. The program has been saved when the display reads **donE**.

You can also store a modified program by pressing and holding the save button at any time. When the display reads **donE**, the 602 reverts to whatever mode it was in when the save button was pressed.

7.5.2 Metering

The 602 has two LED bargraphs that serve as input and output meters. In addition, the right-hand bargraph does double duty as a gain-reduction meter whenever you are editing any of the dynamics group. In gain-reduction mode, the meter indicates the change, from unity gain, for the current function and the LEDs read (and move) from right to left. When operating as a level meter, the LEDs read (and move) from left to right. Each mode has its own scale markings, as shown on the front panel.

Both bargraphs are calibrated as headroom meters. This means that the scale of the meter is referenced to digital clipping (full-scale), and in the case of the output bargraph, digital clipping corresponds to clipping at the analog outputs. Both meters are peak responding. Therefore, adjusting the output level for 6 dB of output headroom sets the output level so that the highest **peak** signal level falls 6 dB below clipping or at +15 dBm (peak) at the balanced output. Now it happens that the peak-to-average ratio for most music falls somewhere between 10 and 20 dB (which means that the peak level ends up being 10 to 20 dB higher than the average, which is what you read on a VU meter). Thus the average level could be anywhere between -5 dBm to +5 dBm (-9 to +1 VU) at the balanced output, depending on the source material.



7.5.3 Gain Setting

There are three places to adjust the gain of the 602: at the analog inputs, before the DSP section, and after the DSP section. An understanding of this topic is essential to getting the most from your 602. A more basic discussion can be found under the heading, "Gain Setting," in Chapter 2.

First, the analog input gains. You make best use of the 602's signal-to-noise ratio by ensuring that your analog input signals are adjusted to just (barely) fit within the input range of the A/D converter. Doing so ensures that the entire conversion range of the converter gets used, ensuring maximum dynamic range through the digital portions of the unit. Set the LINE gain controls so that the 6 dB input headroom LED illuminates on signal peaks. The red CLIP LED should never illuminate.

Next, the input digital gain (**gAIn** in the global parameter group). Most of the time, set the digital input gain to 0 dB. If you are heavily equalizing at 2 or more overlapping frequencies on the equalizer, you may also need to reduce the digital signal level slightly to accommodate the extra boost. If you have a weak analog input signal, and the analog input gain is already wide open, then it is OK to add some digital input gain to bring the overall signal level up. Note that doing so will cost you some noise performance.

Finally, the output digital gain (LEVEL/PAN switch). Set this parameter so that the 2 dB output headroom LED illuminates on signal peaks. The red CLIP LED should never illuminate.

Neither CLIP LED monitors the signal levels within the DSP blocks; you must use your ears If you fear clipping within the DSP blocks, you can always reduce the input digital gain (**gAIn**) slightly.

7.5.4 Equalization

The 602's parametric equalizer has three overlapping bands. Each band can operate as a peaking or notching equalizer, and bands 1 and 3 may be converted to lowpass and highpass shelving curves. Each band operates over a range of 31 to 21.11 kHz with a bandwidth range of .05 octave to 3 octaves. The boost and cut range for each band is +18, -50 dB.

Since each band covers the same frequency range, it is possible to apply equalization at the same frequency in three places. Doing so could conceivably increase the signal level by 54 dB at one frequency. You may need to reduce the input digital gain to avoid distortion.

Electronic considerations aside, one of the contributing factors to an equalizer's sound is its bandwidth. Figure 7-6 lists the bandwidths (octaves) for several (possibly) familiar analog equalizers, as found on their respective mixing consoles. While we make no promise that the 602 will sound identically, these settings may be a good starting point if one of these equalizers is within your frame of reference.[1]

Name	BW (min)	BW (max)
API 550	1.6	n/a
Focusrite	0.6	1.8
Neve V3	0.2	3.0
SSL G	1.4	2.8
SSL E	0.5	2.5

Figure 7-6. Bandwidth Specs for some popular equalizers.

A parametric equalizer offers perhaps the greatest flexibility of any type of equalizer, however it can be more difficult to arrive at a setting than with other equalizers. A good strategy for setting any equalizer is to set the level control for maximum boost, then vary the FREQUENCY and BANDWIDTH until you locate the portion of the spectrum that you wish to modify. Then refine the setting of the LEVEL control for that band. Next refine the setting of the BANDWIDTH control. You may have to go back and forth

[1] The source for these numbers is actual performance graphs published in the following article: *EQ Empirically*, Keith Andrews, Studio Sound magazine, December 1991. The API and Focusrite equalizers were measured at Symetrix.

between LEVEL and BANDWIDTH to find the magic setting. Toggling the band-switch between in and **out** can help too.

It is much easier to hear changes in amplitude (level) than it is to hear bandwidth changes. It is also easier to hear the abundance of something rather than the absence of the same thing. Even if you intend to apply cut (negative level) to a particular frequency, it is still easier to find that frequency by boosting first, tuning second, and resetting the boost/cut last according to taste or need.

It's generally easier to apply boost to a sound for shaping (and that's how many engineers. start). Many times, however, you may want to experiment with removing an offending sound (as opposed to drowning it out with something else). In a complex mix, this may work better because it may require less overall EQ to remove the offending sound; the end result will sound more natural.

7.5.5 Metering and the Dynamics Block

Each component of the dynamics block uses a concept called "gain-reduction." Gain-reduction is the degree to which the overall gain has been lowered in response to some signal condition. When adjusting any of the dynamics block components (dynamic noise reduction, de-esser, expander, compressor or AGC), the right-hand LED meter changes to a gain-reduction meter. Use the lower scale to translate the meter indication into numbers. The meter reverts to displaying level whenever you leave any of the dynamics block.

7.5.6 Dynamic Noise Reduction

The dynamic noise reducer (NR) uses a sliding lowpass filter controlled by the relative level of the signal rejected by the filter. This topology makes a filter that responds more to the content of the signal than its absolute level; it is easier to adjust.

There are three front-panel adjustable parameters: FREQ and THRESH. The FREQ parameter sets the resting frequency of the sliding filter and has a range of 1 kHz to 21.11 kHz. The THRESH parameter sets the relative threshold (**r**) of the onset of filter activity. Pressing on the THRESH switch again accesses the absolute threshold (**A**), which governs the transition between spectral content and signal level as the basis for the filter's action.

In general, use lower resting frequencies to remove excess noise. Higher resting frequencies result in a more subtle action. To set the NR, with signal applied, set the resting FREQuency at 1 kHz. Vary the THRESHold setting until you see activity on the right LED display. Listening, you should hear the noise reduction removing the noise and more than likely your signal. Set the threshold at 0 (zero). Raise the filter FREQuency until you hear onset of the noise. Lower the filter FREQuency until you hear the noise disappear. Now lower the THRESHold setting until you find the magic compromise between the noise, the music, and the audibility of the filter working. Higher THRESHold settings (closer to zero) make it more difficult to "open up" the dynamic filter and lower settings (closer to -35) cause the filter to almost always run "wide open."

Finally, use the absolute THRESHold (**A**) to determine the signal level at which you want the filters action to become level dependent. Usually, this is at a fairly low level, and it is probably more important to eliminate the noise, even at the expense of the signal. The useful range for this parameter runs from -80 dB to -50 dB.

7.5.7 De-Esser

The de-esser uses a limiter controlled by a mildly peaked highpass filter in its sidechain. In sibilant speech, the dominant frequency component is the sibilance itself. Reducing the overall gain during periods of sibilance reduces the level of the sibilant.

In mastering applications, the de-esser can also be useful to reduce excessive high-frequency content, for instance, repairing a mix when the cymbals have too much high frequency content and clutter the high end of the mix.

Set the de-esser by adjusting the THRESH level until the sibilance is no longer objectionable. The de-esser and the noise reduction may be used simultaneously. None of the other dynamics parameters are applicable to the de-esser.

7.5.8 Compression

The compressor generally controls peak levels and maintains a high overall average signal level. Used in this manner, the compressor's action is generally inaudible. Compressors can also be used creatively, to make a source sound louder than it really is, or to create a special effect.

For most level control applications, moderate settings yield the best results. We recommend a starting point of: THRESH setting sufficient to cause about 6 to 8 dB of gain reduction on peaks using a RATIO setting of 4:1. Pick an ATTACK time that allows enough of the initial sound through to not lose crispness, and a RELEASE time that allows the compressor to partially recover (gain reduction display almost out) between words.

For a highly compressed sound (you know, the used car salesman during the 3AM movie), use a 10:1 ratio setting, 10 dB or more of gain reduction, and a fast release time (fast enough to cause breathing).

7.5.9 AGC

An AGC (Automatic Gain Control) is simply a smart compressor that knows when to allow its gain to change. This simple concept allows using a compressor to track a varying audio signal while maintaining a more constant output level. Note that the goal is to reduce the overall variation in signal level, not to remove all variation completely.

You set the AGC much like you set the 602's compressor. The big difference is the THRESH setting, which becomes the auto-release threshold. This determines the level at which the compressor allows its gain to rise. (You don't want the gain to rise trying to track a signal buried in noise, right?) Set the THRESH so that the lowest desired signal causes fluctuation in the gain reduction meter.

7.5.10 Downward Expander

The downward expander reduces its gain for any signal level below the threshold setting. Typically, downward expanders are used to remove noise or unwanted signal from an audio signal by simply lowering the gain when the overall level falls below threshold.

Think about using the expander when you are faced with a noisy signal (not necessarily hiss) or when heavily compressing a voice and you want to remove some of the less desirable artifacts (false teeth rattling, lip smacking, tongue noise, etc.) You can also use the expander to help remove microphone leakage from a signal.

Start by setting the expansion RATIO to 1:2. This means that the output falls 2 dB for every 1 dB of below-threshold change in the input signal. Next set the threshold so that the expander causes gain reduction (right LED meter) as the signal falls in level. Higher expansion ratios will make the effect more obvious. The ATTACK parameter determines the expander's response to a signal's duration; shorter attack times allow the expander to respond to short-duration sounds (like clicking your tongue). If the attack time is long enough, the expander will ignore short-duration sounds.

The RELEASE time parameter determines the length of time needed for the gain to drop once the input signal abruptly falls below threshold. The RELEASE time and the expansion RATIO appear to interact somewhat. This is not the case. A 1:8 expansion ratio means that the output level

will fall by 8 dB for every 1 dB of input change below threshold. A release time of 1000 milliseconds says that it will take 1000 milliseconds for the output signal to decay from its initial value. The RATIO parameter deals with the slope of the input vs output gain relationship, independent of time and the RELEASE parameter deals with the rate-of-change (in time units) of the output signal when it transits the two points (initial attenuation and ultimate attenuation) as determined by the ratio setting.

7.5.11 Delay

The delay section of the 602 uses two delay lines having separate inputs and separate outputs. The outputs drive a lowpass filter that feeds the output mix and the feedback controls. Each feedback signal mixes with the input signal at the delay line input of the opposite channel (the feedback is cross-coupled). A signal flow diagram may be found in Figure 7-3.

The delay times of the delays may be adjusted independently or ganged together. The feedback factor, lowpass filter frequency, delay time rate-of-change, level-related rate-of-change, and the wet-dry mix are independently adjustable. Finally, the delay time of the two delays may be modulated with the rate, waveform, and depth parameters being adjustable.

The delay modulation source is either a sine-wave generator, triangle-wave generator or a random number generator. The RATE parameter sets either the sine/triangle-wave frequency. or the random number generator's update rate. The depth control limits the range of the delay time modulation. Holding down the RATE button changes the delay modulation source.

All of the previously mentioned parameters may be programmed via MIDI.

7.5.11.1 Echo effects

Creating an echo consists of delaying the input signal by some amount, then adding the delayed signal back to itself. This creates an echo having one repeat. To create this type of sound, set the MIX to 50%, set the DELAY to 330 ms for both channels, set the FILTER to 18 kHz, and finally set the FEEDBACK to 0. You (into mic): "Hello." 602: "Hello Hello." Experiment with different delay times. What does it sound like when the delay time is quite short, say around 10 ms? What does it sound like when the delay time is mid-range, say 40 to 80 ms. Now experiment with different mix settings. Listen in stereo and make the two delay times slightly different. Now try making them radically different. Try using **duAL** mode to sweep the different delay times.

You create repeating echoes by recirculating the output of the delay line back to its input. On the 602 set the FEEDBACK to **P-10**, set the DELAY time to 330 ms. Now speak into the mic. You: "Hello." 602: "Hello Hello Hello Hello Hello Hello ..." Higher feedback settings increase the number (and duration) of the echoes. Be sure that you try varying the wet/dry mix as well as the feedback and delay times.

7.5.11.2 Flanging

It's Audio history time. The term "flanging" came about because the effect was originally created by using two three-head tape recorders (30 years ago, that was how we created delay), inputs paralleled, outputs mixed. Then the engineer held his thumb on the reel flange of one machine to slow it down slightly (which changed the time delay). Varying the pressure on the reel flange changes the effect. That's more or less what happened when The Small Faces made "Itchykoo Park," about 25 years ago.

Flanging is nothing more than comb filtering. The modulation oscillator replaces the thumb on the reel flange. On the 602, you create flanging by choosing a very short DELAY time, 0.5 to 2 milliseconds, set the MIX at 50%, set the modulation RATE at 1, set the modulation DEPTH at 100. You should hear a hollowness (the jet plane sound) that changes with time. Increasing the amount of FEEDBACK makes the effect more pronounced. Changing the FEEDBACK polarity/phase shifts the comb frequencies. Increasing the delay-time Rate-of-change (**rt**) by holding down the DELAY button until the display reads **rt** smooths out the transitions and

makes the changes smoother. You should definitely try making the two delay times slightly different. Be sure to experiment with the different modulation sources.

7.5.11.3 Chorus effects

Chorusing is a variation on flanging. The effect gives the impression of multiple sources. On the 602, start with the DELAY time at about 10 milliseconds, MIX at 50%, FEEDBACK around 80, modulation RATE around 20, modulation DEPTH at 100, and the delay-time rate-of-change at its minimum setting. Listen in stereo.

Experiments should include varying the delay time(s), altering the rate-of-change, altering the wet/dry mix, and the modulation parameters.

Notes

8. Applications

Here are a few applications that the 602 lends itself to. Do you have an unusual application for the 602? Send it to us and we'll consider sending you a can of slug chowder or some chocolate covered espresso beans from Starbucks for your trouble (novelness of idea limited to our opinion, decision of judges is final, offer void where taxed or prohibited).

8.1 Broadcast Voice Processing

Use the 602 to create a unique sound for each of your on-air personalities. Give each announcer his or her own program number, then create and store their sound. If you have a way to send MIDI information to the 602 under control of a clock, then the announce mic processing can change at shift-change time. Connect the 602 as an insert device after your console's mic preamp.

8.2 Voice-over Processing

Create and store each of your favorite voice-talent's settings in the 602. The next time that you work with them, your starting point is a button press away.

8.3 Foley Processing

Use the 602 as in insert device in the console's signal path. You can also use it to process field tapes during transfer to a workstation or other storage system.

8.4 Digital Mastering

Since the 602 can operate in digital-in/digital-out mode, use it when making production masters of digital material. You can add compression, make level changes, EQ changes, etc.

If you're not particularly enamored with the ADC in your digital recorder, you can use the 602 as a converter if your digital recorder has digital (S/PDIF or AES/EBU) inputs.

8.5 Musical Applications

The particular combination of processors in the 602 make it ideal as an instrument processor, especially for electronic keyboards. In the studio, you could go as far as to use the AES/EBU outputs as a direct-digital output while listening to the analog outputs.

8.6 Sound Reinforcement Applications

One possible sound-reinforcement application for the 602 is that of an ultimate channel insert processor. Just think, one channel insert patch and you have a parametric equalizer, de-esser, de-noiser, compressor, AGC, and stereo delay at your fingertips. Use just one, use them all, the important thing is that they're all there.

Another application simply uses the delay and possibly the EQ as an equalized stereo delay line. The simplified user-interface makes parameter changing fast and easy, the programmability helps make changing modes easy.

Notes

9. Troubleshooting Chart

Symptom	Probable Cause
No output	Check cables and connections. Are inputs driven by outputs, and outputs driving inputs? Verify cables, source and load by patching input and output connections together, at the unit. Check for AC power presence. Check output by plugging headphones into analog output connector (use an adapter). Are the HEADROOM displays operating?
Hum or buzz in output	Check input and output connector wiring (refer to Figure 3.3). Ground loop. check related system equipment grounding. Are all system components on the **same** AC ground?
Distortion	Check input signal. Is it too hot, or is it already distorted? Is the HEADROOM display indicating clipping? Check output loading. Should be above 600 ohms. Are the power amplifier(s) clipping? Is something else clipping? Check input digital gain and output digital gain settings.
Noise (hiss)	Check input signal levels, and level control setting. The HEADROOM display should indicate signal, up to but not including the CLIP led. Check gain settings on downstream equipment. The system gain structure should be such that the 602 operates at or near unity gain. Is the input signal already noisy?
No LED display	Is the unit plugged in, and turned on? Is the AC outlet OK?
No nothing	Is the unit in BYPASS mode?
Display reads '**Er nn**' (nn is a two digit number).	Power up error. Try turning the unit off, then on again. Write the number down before you call us.
Display flashes ' **bAt**' at turn-on.	Memory backup battery death throes. You have about two weeks to replace the battery before you lose your programs. Contact the factory before trying to do this yourself.
Unit not plugged in, but works anyway	Call us.

Notes

10. 602 Stereo Digital Processor Limited Warranty

This Symetrix product is designed and manufactured for use in professional and studio audio systems. Symetrix, Inc. (Symetrix) warrants that this product, manufactured by Symetrix, when properly installed, used, and maintained in accordance with the instructions contained in the product's operator's manual, will perform according to the specifications set forth in the operator's manual.

Symetrix expressly warrants that the product will be free from defects in material and workmanship for one (1) year. Symetrix' obligations under this warranty will be limited to repairing or replacing, at Symetrix' option, the part or parts of the product which prove defective in material or workmanship within one (1) year from date of purchase, provided that the Buyer gives Symetrix prompt notice of any defect or failure and satisfactory proof thereof. Products may be returned by Buyer only after a Return Authorization number (RA) has been obtained from Symetrix and Buyer will prepay all freight charges to return any products to the Symetrix factory. Symetrix reserves the right to inspect any products which may be the subject of any warranty claim before repair or replacement is carried out. Symetrix may, at its option, require proof of the original date of purchase (dated copy of original retail dealer's invoice). Final determination of warranty coverage lies solely with Symetrix. Products repaired under warranty will be returned freight prepaid via United Parcel Service by Symetrix, to any location within the Continental United States. Outside the Continental United States, products will be returned freight collect.

The foregoing warranties are in lieu of all other warranties, whether oral, written, express, implied or statutory. Symetrix, expressly disclaims any IMPLIED warranties, including fitness for a particular purpose or merchantability. Symetrix's warranty obligation and buyer's remedies hereunder are SOLELY and exclusively as stated herein.

This Symetrix product is designed and manufactured for use in professional and studio audio systems and is not intended for other usage. With respect to products purchased by consumers for personal, family, or household use, Symetrix **expressly disclaims all implied warranties, including but not limited to warranties of merchantability and fitness for a particular purpose.**

This limited warranty, with all terms, conditions and disclaimers set forth herein, shall extend to the original purchaser and anyone who purchases the product within the specified warranty period.

Warranty Registration must be completed and mailed to Symetrix within thirty (30) days of the date of purchase.

Symetrix does not authorize any third party, including any dealer or sales representative, to assume any liability or make any additional warranties or representation regarding this product information on behalf of Symetrix.

This limited warranty gives the buyer certain rights. You may have additional rights provided by applicable law.

Limitation of Liability

The total liability of Symetrix on any claim, whether in contract, tort (including negligence) or otherwise arising out of, connected with, or resulting from the manufacture, sale, delivery, resale, repair, replacement or use of any product will not exceed the price allocable to the product or any part thereof which gives rise to the claim. In no event will Symetrix be liable for any incidental or consequential damages including but not limited to damage for loss of revenue, cost of capital, claims of customers for service interruptions or failure to supply, and

costs and expenses incurred in connection with labor, overhead, transportation, installation or removal of products or substitute facilities or supply houses.

11. Repair Information

Should you decide to return your 602 to Symetrix for service, please follow the following instructions.

11.1 Return Authorization

Symetrix will service any of its products for a period of five years from the date of manufacture. However, no goods will be accepted without a Return Authorization number.

Before sending anything to Symetrix, call us for an RA number. just ask, we'll gladly give you one! call (206) 787-3222, weekdays, 8am to 4:30 pm pacific time.

11.2 In-Warranty Repairs

To get your unit repaired under the terms of the warranty:

1. Call us for an RA number.

2. Pack the unit in its original packaging materials.

3. Include your name, address, etc. and a brief statement of the problem. Your daytime telephone number is very useful if we can't duplicate your problem.

4. Put the RA number on the outside of the box.

5. Ship the unit to Symetrix, freight prepaid.

Just do those five things, and repairs made in-warranty will cost you only the one-way freight fee. We'll pay the return freight.

If you choose to send us your product in some sort of flimsy, non-Symetrix packaging, we'll have to charge you for proper shipping materials. If you don't have the factory packaging materials, then do yourself a favor by using an oversize carton, wrap the unit in a plastic bag, and surround it with bubble-wrap. Pack the box full of Styrofoam peanuts. Use additional bubble-wrap if you must ship more than one unit per carton. Be sure there is enough clearance in the carton to protect the rack ears (you wouldn't believe how many units we see here with bent ears). We won't return the unit in anything but original Symetrix packaging. Of course, if the problem turns out to be operator inflicted, you'll have to pay for both parts and labor. In any event, if there are charges for the repair costs, you will pay for return freight. All charges will be COD unless you have made other arrangements (prepaid, Visa or Mastercard).

11.3 Out-of-Warranty Repairs

If the warranty period has passed, you'll be billed for all necessary parts, labor, packaging materials, and any applicable freight charges.

Remember, you must call for an RA number before you send the unit to Symetrix.

Notes

12. Specifications

Input/Output

Analog Inputs	XLR-female, 12.5-kilohms line-level balanced bridging.
Digital Inputs	Two, XLR-female and RCA Female, AES/EBU or S/PDIF
Analog Outputs	Two, 300-ohm source impedance, balanced. XLR-male
Digital Outputs	Two, XLR-male and RCA Female, AES/EBU or S/PDIF
Maximum input level	+22 dBu
Maximum output level	+21.5 dBu

Filter Block

Type	Three-band parametric equalizer
Shelving Characteristic	31 Hz to 21.11 kHz, Baxandall approximation
Peak/Dip Bandwidth	0.05 to 3 octaves
Maximum boost/cut	18 dB boost, -50 dB cut

Delay Block

Effects	Echo generation with filtered feedback, distance simulation, flanging, or chorusing.
Delay time	0.5 ms to 330 ms
Lowpass frequency	600-18 kHz
Modulation	Random, sine-wave, or triangle-wave
Depth	0-100%

Dynamics Block

Types	De-essing, dynamic noise reduction, downward expansion, compression, AGC/leveling.
Compression ratio (max)	10:1
Expansion ratio (max)	1:8
Attack time	100 microseconds to 10,000 ms
Release time	100 ms to 10,000 ms
DS (De-Ess)	High-ratio limiter driven by sibilance content.
NR (dynamic noise reduction)	Sliding low-pass filter driven by high-frequency energy content.

Output Processing

Types	Level and pan

Performance Data

Frequency Response	12 Hz - 20 kHz +/- 1.5 dB
Distortion (THD)	< .01% @ 1 kHz, 1V RMS
Dynamic Range	>104 dB. This represents the difference between the largest and smallest signals that will pass through the 602. Measured using 8192 point FFT with Blackman-Harris windowing function.
Sample Rates	44.1 kHz, 48kHz
Converter Type	Delta-Sigma
Conversion method	18-bit linear, 64X oversampling
Parameter Storage	RAM with battery backup
Group Delay	1.4-3.98 ms @ 48 kHz, 1.51-4.11 ms @ 44.1 kHz
Input Headroom Display	9-LED bargraph
Analog Input Clip Indicator	Red LED indicates clipping at analog inputs.
Output Headroom Display	8-LED bargraph

Midi Implementation

Access	MIDI program change, sysex, aftertouch, pitch bend, bank select
MIDI channel range	1-128, omni-mode
Accessible parameters	Most front panel parameters plus internal constants
Connectors	midi in, midi out
Data dump	current program or entire memory
Manufacturer ID	00, 00, 5e

Physical

Size (hwd), in & cm	1.75 x 19 x 7 in
	4.44 x 48.26 x 17.78 cm
Weight, lbs & kg	7.6 lbs (3.5kg) net 10 lbs (4.6kg) shipping

Electrical

Power requirements	117V AC nominal, 105-125V ac 50-60 Hz, 20 watts 230V AC nominal, 205-253V ac 50 Hz, 20 watts.

In the interest of continuous product improvement, Symetrix Inc. reserves the right to alter, change, or modify these specifications without prior notice.

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A. Editing Realtime Midi Settings

The 602 has the capability to modify its parameter settings in realtime, either as a function of one of the MIDI continuous controllers or from an internal control source. To access the realtime MIDI settings from the front panel, press the MIDI button until the display reads **rEAL**. A long press on the MIDI button then accesses the realtime block editor. Successive presses of the MIDI button access each item on the realtime MIDI submenu. Each submenu item allows editing one of the realtime MIDI parameters.

Each menu item is described as follows:

bLC Block select MIDI linkage **0**, **1**, or **SEt**. These linkages represent the two available realtime MIDI setups. This menu item selects either the first or second MIDI linkage for editing or selects an arbitrary parameter (offsets 0-70) within the edit buffer for setting.

PAr Selects the edit buffer parameter to be edited. Use the edit buffer table to look up the offset of the desired parameter. The right-hand bargraph (output headroom/gain reduction) shows the parameter's current actual edit buffer value on a linear scale of 0 (no LEDs) to 127 (all LEDs).

SrC Selects the source of the realtime MIDI control, as follows:

Display	Description
oFF	off
Cn	MIDI control change packet (needs 2nd parameter setting)
AF	MIDI after touch
Pb	MIDI pitch bend
dL1	Delay modulation oscillator 1
dL2	Delay modulation oscillator 2
LoG	Dynamics section log signal level
nr	NR center frequency (NR must be engaged)
CGr	Instantaneous compressor gain reduction
EGr	Instantaneous expander gain reduction
bLC1	Block 1 output (only in Block 2 edit)

2nd MIDI control type for MIDI control change packet.

SCAL Scaling factor to apply to source after adding offset value (±4).

oFt Offset to apply to source value.

CLPL Lowest value allowed from this block, after all processing, scaling, and offset.

CLPH Highest value allowed from this block, after all processing, scaling, and offset.

Press the LEAVE EDIT button to exit the realtime MIDI editing mode. if the **SrC** or **oFt** parameters were modified, then the menu item temporarily reverts to the previous parameter (**PAr** or **SCAL**). Changing the **PAr** parameter temporarily disables the parameter update for a second. This helps avoid accidentally (and worse, invisibly) overwriting another edit buffer entry while selecting a new one. Be careful! Mysterious things (not necessarily wonderful) can happen when the edit buffer values are arbitrarily (or randomly) rewritten...

To restore a program after clobbering the edit buffer, reload the source program over the existing edit buffer or overwrite the edit buffer with ROM program 256, then write the edit buffer over the zombie program's memory location.

When selecting one of the realtime blocks, you have the third option of selecting **SEt**. This acceesses any parameter within the edit buffer, displays its value, and allows you to change that value. SEt has the following menu items:

PAr Selects the edit buffer parameter to be edited. Use the edit buffer table (Appendix C) to look up the offset of the desired parameter. The right-hand bargraph (output headroom/gain reduction) shows the parameter's current actual edit buffer value on a linear scale of 0 (no LEDs) to 127 (all LEDs).

oFt Offset (from zero) to apply to source value.

Press LEAVE EDIT to exit this mode.

A.1 Realtime MIDI Example.

Refer to the description of Realtime MIDI Block 1, which can be found in Appendix C. We'll use modulation source six, the dynamics section log signal level. This control source represents the logarithm of the signal level presented to the dynamics section. The dynamics section uses it to drive the compressor, expander, and AGC sidechains. We'll use it to control the filter frequency of band 2 of the parametric equalizer block.

Begin by accessing band 2 of the equalizer. Set the LEVEL parameter to +15 dB, and the BW parameter to 0.5.

Access realtime edit mode by pressing the MIDI button until the display reads **rEAL**.

Pressing the MIDI button again, the display now reads **bLC**. Select block **0** using the Wheel.

Press the MIDI button again. Use the Edit Buffer parameter table to locate the offset of the desired parameter. We want parameter **10**.

Press the MIDI button again. The display reads **SrC**. Since we want the log signal level to become the controller for the filter frequency, select **6** as the modulation source using the Wheel.

Press the MIDI button twice. The display should now read **oFt**. While listening to the 602's output, adjust the offset value until you hear the equalizer filter begin to work. You can then use the **SCAL** parameter to alter the range of the effect.

Note: whenever one of the realtime blocks has been set up and attached to a parameter, trying to edit that same parameter from the front panel results in the message **rEAL** whenever the Wheel is turned. This is true only for realtime blocks one and two, and not for the **SEt** function. Accessing the parameter attached to the realtime block, from the front panel, results in the display temporarily showing the current value of the parameter. This can be useful in determining useful limits for a parameter, or to view the results of offset and scale operations.

B. Using the Lexicon MRC to Edit Realtime MIDI Settings

Most of the 602's internal parameters may be modified remotely using MIDI. In addition, many parameters may be modified dynamically, while the 602 is passing signal (as if there was a front panel knob(s) for the parameter(s)). The Lexicon MRC (MIDI Remote Controller) can be used to edit the internal dynamic midi settings in the 602 by following these steps. Although this chapter is devoted to the MRC and its use with the 602, there are other MIDI controllers available that can perform comparably. This procedure was developed on an MRC having software revision 3.01.

This procedure uses machine 15, MIDI port 1, setup 9 to control the delay block and setup 10 to control the real-time MIDI block. The following table lists the steps needed to program the MRC to accomplish this. We assume that you already own a MRC and are somewhat familiar with how it works. The table is divided into steps and each step has four parts: MRC Key, MRC Display, Data to Set/Enter, and Comments.

❒ MRC Key represents a key on the MRC that you must press.

❒ MRC Display represents the display on the MRC. In some cases, this represents a portion of the MRC display (for instance, a label for one of the sliders).

❒ Data to Set/Enter represents data that you must enter into the MRC via the sliders, buttons, or keypad.

❒ Comments are just that: comments.

The following 9 steps exactly parallel the first 9 steps of the table. Refer also to the edit buffer parameter tables in Appendix C. Read the notes presented after the procedure. They explain some of the details behind the steps. This should help if you're trying to translate the procedure to a different MIDI controller.

Step	What to do
1	Press the MACH key on the MRC. The display reads "MACH # ..." Press 15 on the keypad, or use slider 1 to set the machine number to 15.
2	Press the ENTER key on the MRC
3	Press the SETUP key on the MRC. The display reads "SOURCE DEST ..." Press 1 on the keypad, or use slider 1 to ensure that the current setup is 10.
4	Press the ENTER key on the MRC.
5	Press the EDIT key on the MRC.
6	Adjust the slider under the "SOURCE" prompt on the display so that the source is "slr1."
7	Adjust the slider under the "DEST" prompt so that the DEST is "SYSEX."
8	Adjust the slider under the "OUT#" prompt so that the OUT# is 1.
9	Press the PAGE key.

Note: The length of this example may, at first, make the whole thing seem daunting or extremely complicated. It is not. A vast majority of the process is highly repetitive and once you have programmed two or three sliders, the pattern should begin to emerge and you can begin working from memory.

Note: Whenever one of the realtime blocks has been set up and attached to a parameter, trying to edit that same parameter from the front panel results in the message **rEAL**, whenever the Wheel is turned. This is true only for realtime blocks one and two, and not the **SEt** function under realtime MIDI edit.

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
1	MACH	MACH #…	15	set to machine 15
2	ENTER			
3	SETUP	GMIDI SETUP#	10	set to setup 10 via slider 1 or keypad
4	ENTER			
5	EDIT	SOURCE DEST …		now edit the setup
Make slider 1 control the event type (see Real time MIDI Block 1)				
6		SOURCE	slr1	source to slider 1
7		DEST	sysex	dest to SYSEX
8		OUT#	1	midi out #1
9	page	DEFINE SYSEX BYTES 2-3		next page
10			F0	byte 1, can't change (midi SYSEX)
11			00	mfrID0
12			00	mfrID1
13	page	DEFINE SYSEX BYTES 4-7		next page
14			5E	mfrID2
15			01	device type
16			00	unit/channel
17			1C	edit buffer data set
18	page	DEFINE SYSEX BYTES 8-9		
19			47	edit buffer 71
20			BYTE	send slider setting
21	page	LABEL FOR slr1	T Y P E	use sliders 1-4 to set label to "T Y P E"
Make slider 2 control the second parameter of the Real time MIDI block				
22	page	SOURCE DEST OUT#		**now do slider 2**
23		SOURCE	slr2	use button 1 to change source
24		DEST	SYSEX	use slider 2 to set dest to SYSEX
25	page	DEFINE SYSEX BYTES 2-3		next page
26			F0	SYSEX
27			00	mfrID0
28			00	mfrID1
29	page	DEFINE SYSEX BYTES 4-7		next page
30			5E	mfrID2
31			01	device type
32			00	unit/channel
33			1C	edit buffer data set
34	page	DEFINE SYSEX BYTES 8-9		next page
35			48	edit buffer 72
36			BYTE	send slider setting
37	page	LABEL FOR slr2		next page

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
38			2 N D P	use sliders 1-4 to set label to "2 N D P"
Make slider 3 control the offset applied to slider 1's source				
39	page	SOURCE DEST OUT#		**setup for slider 3**
40		SOURCE	slr3	use button 1 to set source to slr3
41		DEST	SYSEX	use slider 2 to set DEST to SYSEX
42		OUT#	1	use slider 3 to set OUT# to 1
43	page	DEFINE SYSEX BYTES 2-3		next page
44			F0	SYSEX
45			00	mfrID0
46			00	mfrID1
47	page	DEFINE SYSEX BYTES 4-7		
48			5E	mfrID2
49			01	device type
50			00	unit/channel
51			1C	edit buffer data set
52	page	DEFINE SYSEX BYTES 8-9		
53			49	edit buffer 73
54			BYTE	send slider setting
55	page	LABEL FOR slr3		
56			O F F S	use sliders 1-4 to set label to "O F F S"
Make slider 4 control the multiplier value				
57	page	SOURCE DEST OUT#		**setup for slider 4**
58		SOURCE	slr4	use button 1 to set source to slr4
59		DEST	SYSEX	use slider 2 to set DEST to SYSEX
60		OUT#	1	use slider 3 to set OUT# to 1
61	page	DEFINE SYSEX BYTES 2-3		
62			F0	SYSEX
63			00	mfrID0
64			00	mfrID1
65	page	DEFINE SYSEX BYTES 4-7		
66			5E	mfrID2
67			01	device type
68			00	unit/channel
69			1C	edit buffer data set
70	page	DEFINE SYSEX BYTES 8-9		
71			4A	edit buffer 74
72			BYTE	send slider setting
73	page	LABEL FOR slr2		

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
74			M P L Y	use sliders 1-4 to set label to "M P L Y"
Slider 5 determines the parameter affected by the event selected by slider 1.				
75	page	SOURCE DEST OUT#		**setup slider 5**
76		SOURCE	slr5	use button 1 to set source to slr5
77		DEST	SYSEX	use slider 2 to set DEST to SYSEX
78		OUT#	1	use slider 3 to set OUT# to 1
79	page	DEFINE SYSEX BYTES 2-3		
80			F0	SYSEX
81			00	mfrID0
82			00	mfrID1
83	page	DEFINE SYSEX BYTES 4-7		
84			5E	mfrID2
85			01	device type
86			00	unit/channel
87			1C	edit buffer data set
88	page	DEFINE SYSEX BYTES 8-9		
89			4F	edit buffer 79
90			BYTE	send slider setting
91	page	LABEL FOR slr2		
92			P A R M	use sliders 1-4 to set label to "P A R M"
93	page	SOURCE DEST OUT#		
94	store		1	**save your work!**
Use machine 15, setup 9 to control the delay.				
95	mach	MACH # 15 GMIDI Setup 1	16	use slider 1 to set mach# to 15
96	enter	TYPE 2NDP OFFS MPLY		
97	setup	GMIDI SETUP# 1 DYNAM 1	9	
98	enter	TYPE 2NDP OFFS MPLY		
99	edit	SOURCE DEST OUT#		
Set slider 1 to control delay 1				
100		SOURCE	slr1	use button 1 to set source to slr1
101		DEST	SYSEX	use slider 2 to set DEST to SYSEX
102		OUT#	1	use slider 3 to set OUT# to 1
103	page	DEFINE SYSEX BYTES 2-3		
104			F0	SYSEX
105			00	mfrID0
106			00	mfrID1
107	page	DEFINE SYSEX BYTES 4-7		

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
108			5E	mfrID2
109			01	device type
110			00	unit/channel
111			1C	edit buffer data set
112	page	DEFINE SYSEX BYTES 8-9		
113			22	edit buffer 34
114			BYTE	send slider setting
115	page	LABEL FOR slr1		
116			D L Y 1	use sliders 1-4 to set label to "D L Y 1"
Set slider 2 to control delay 2				
117	page	SOURCE DEST OUT#		**setup slider 2**
118		SOURCE	slr2	use button 1 to set source to slr2
119		DEST	SYSEX	use slider 2 to set DEST to SYSEX
120		OUT#	1	use slider 3 to set OUT# to 1
121	page	DEFINE SYSEX BYTES 2-3		
122			F0	SYSEX
123			00	mfrID0
124			00	mfrID1
125	page	DEFINE SYSEX BYTES 4-7		
126			5E	mfrID2
127			01	device type
128			00	unit/channel
129			1C	edit buffer data set
130	page	DEFINE SYSEX BYTES 8-9		
131			23	edit buffer 35
132			BYTE	send slider setting
133	page	LABEL FOR slr2		
134			D L Y 2	use sliders 1-4 to set label to "D L Y 2"
135	page	SOURCE DEST OUT#		
Set slider 3 to control feedback (recirculation).				
136		SOURCE	slr3	use button 1 to set source to slr3
137		DEST	SYSEX	use slider 2 to set DEST to SYSEX
138		OUT#	1	use slider 3 to set OUT# to 1
139	page	DEFINE SYSEX BYTES 2-3		
140			F0	SYSEX
141			00	mfrID0
142			00	mfrID1
143	page	DEFINE SYSEX BYTES 4-7		
144			5E	mfrID2

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
145			01	device type
146			00	unit/channel
147			1C	edit buffer data set
148	page	DEFINE SYSEX BYTES 8-9		
149			25	edit buffer 37
150			BYTE	send slider setting
151	page	LABEL FOR slr3		
152			F B	use sliders 1-4 to set label to "F B "
Set slider 4 to control the wet/dry mix.				
153	page	SOURCE DEST OUT#		**setup slider 4**
154		SOURCE	slr4	use button 1 to set source to slr4
155		DEST	SYSEX	use slider 2 to set DEST to SYSEX
156		OUT#	1	use slider 3 to set OUT# to 1
157	page	DEFINE SYSEX BYTES 2-3		
158			F0	SYSEX
159			00	mfrID0
160			00	mfrID1
161	page	DEFINE SYSEX BYTES 4-7		
162			5E	mfrID2
163			01	device type
164			00	unit/channel
165			1C	edit buffer data set
166	page	DEFINE SYSEX BYTES 8-9		
167			27	edit buffer 39
168			BYTE	send slider setting
169	page	LABEL FOR slr4		
170			M I X	use sliders 1-4 to set label to "M I X"
Set slider 5 to control rate-of-change.				
171	page	SOURCE DEST OUT#		**setup slider 5**
172		SOURCE	slr5	use button 1 to set source to slr5
173		DEST	SYSEX	use slider 2 to set DEST to SYSEX
174		OUT#	1	use slider 3 to set OUT# to 1
175	page	DEFINE SYSEX BYTES 2-3		
176			F0	SYSEX
177			00	mfrID0
178			00	mfrID1
179	page	DEFINE SYSEX BYTES 4-7		
180			5E	mfrID2

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
181			01	device type
182			00	unit/channel
183			1C	edit buffer data set
184	page	DEFINE SYSEX BYTES 8-9		
185			24	edit buffer 36
186			BYTE	send slider setting
187	page	LABEL FOR slr5		
188			R O C	use sliders 1-4 to set label to "R O C"
189	page			
Set switch 1 to set the mix to 0				
190	page	SOURCE DEST OUT#		**setup switch 1**
191		SOURCE	swt1	use button 1 to set source to swt1
192		DEST	SYSEX	use slider 2 to set DEST to SYSEX
193		OUT#	1	use slider 3 to set OUT# to 1
194	page	DEFINE SYSEX BYTES 2-3		
195			F0	SYSEX
196			00	mfrID0
197			00	mfrID1
198	page	DEFINE SYSEX BYTES 4-7		
199			5E	mfrID2
200			01	device type
201			00	unit/channel
202			1C	edit buffer data set
203	page	DEFINE SYSEX BYTES 8-9		
204			27	edit buffer 39
205			0	kills wet signal
206	page	LABEL FOR slr5		
207			K I L L	use sliders 1-4 to set label to "K I L L"
208	page			
Set switch 2 to turn on sine modulation				
209	page	SOURCE DEST OUT#		**setup switch 2**
210		SOURCE	swt2	use button 1 to set source to swt2
211		DEST	SYSEX	use slider 2 to set DEST to SYSEX
212		OUT#	1	use slider 3 to set OUT# to 1
213	page	DEFINE SYSEX BYTES 2-3		
214			F0	SYSEX
215			00	mfrID0
216			00	mfrID1
217	page	DEFINE SYSEX BYTES 4-7		

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
218			5E	mfrID2
219			220	device type
221			222	unit/channel
223			1C	edit buffer data set
224	page	DEFINE SYSEX BYTES 8-9		
225			2A	edit buffer 42
226			227	228 = sine
229	page	LABEL FOR swt2		
230			S I N E	use sliders 1-4 to set label to "S I N E"
231	page			
Set switch 3 to set triangle modulation.				
232	page	SOURCE DEST OUT#		**setup slider 5**
233		SOURCE	slr5	use button 1 to set source to slr5
234		DEST	SYSEX	use slider 2 to set DEST to SYSEX
235		OUT#	236	use slider 3 to set OUT# to 1
237	page	DEFINE SYSEX BYTES 2-3		
238			F0	SYSEX
239			240	mfrID0
241			242	mfrID1
243	page	DEFINE SYSEX BYTES 4-7		
244			5E	mfrID2
245			246	device type
247			248	unit/channel
249			1C	edit buffer data set
250	page	DEFINE SYSEX BYTES 8-9		
251			2A	edit buffer 42
252			7F	7F = triangle
253	page	LABEL FOR swt3		
254			T R I	use sliders 1-4 to set label to "T R I"
255	page			
Set switch 4 to turn on random modulation				
256	page	SOURCE DEST OUT#		**setup switch 4**
257		SOURCE	swt4	use button 1 to set source to swt4
258		DEST	SYSEX	use slider 2 to set DEST to SYSEX
259		OUT#	1	use slider 3 to set OUT# to 1
260	page	DEFINE SYSEX BYTES 2-3		
261			F0	SYSEX
262			00	mfrID0
263			00	mfrID1

STEP	MRC Key	MRC Displays	Data to Set/Enter	Comments
264	page	DEFINE SYSEX BYTES 4-7		
265			5E	mfrID2
266			01	device type
267			00	unit/channel
268			1C	edit buffer data set
269	page	DEFINE SYSEX BYTES 8-9		
270			2A	edit buffer 42
271			0	0 = random
272	page	LABEL FOR swt4		
273			R A N D	use sliders 1-4 to set label to "R A N D"
274	page			
275	store		1	**save your work!**
				you're done.

Notes:

If you take the time to key in this program into your MRC, here's what you'll get:

Setup 9: Delay Block Controller			
SW 1	SW 2	SW 3	SW 4
kill	sine	triangle	random
SLIDER 1	SLIDER 2	SLIDER 3	SLIDER 4
delay1	delay2	feedback	mix
SLIDER 5	SLIDER 6	SLIDER 7	SLIDER 8
ROC	filter	speed	depth

Setup 10: Real Time MIDI Controller			
SW 1	SW 2	SW 3	SW 4
n/a	n/a	n/a	n/a
SLIDER 1	SLIDER 2	SLIDER 3	SLIDER 4
Event type select	2nd parm	offset	scale
SLIDER 5	SLIDER 6	SLIDER 7	SLIDER 8
parm	n/a	n/a	n/a

Step 6. This tells the MRC that the slider that we want to program is slider 1.

Steps 7-18. These steps are the same for every slider and button that we are programming in this example.

Step 19. This number (47h) comes from Appendix C under the heading "Data Structure Per Program." This data structure is a table of offsets, each of which represents one parameter in the edit buffer. Parameter 47h or 71d (47 hex equals 71 decimal) is the Event Type under Real time MIDI Block 1. When the MRC sends the MIDI command represented by steps 10 through 20, the value sent at step 19 tells the 602 that the next data byte received gets stuffed into the edit buffer at offset 71, which is the Real time MIDI Event Type.

Step 20. This step has a value of BYTE, which is the value represented by that slider's setting. This is how you send a slider value to the 602 from the MRC.

Step 21. This step labels the slider so its function is a bit more obvious to humans.

Step 171. This step programs one of the switches (buttons) on the MRC. The buttons are a little different than programming the sliders in that they only send one MIDI message per press, and there is no way to create an ON/OFF toggle on one button. Instead, you must program one button to send an OFF command, and the other button to send an ON command.

Step 172. Notice that we set the source to switch 1 instead of slider 1.

Step 185. Edit buffer offset 27h is the delay block mix control. Sending a 0 value to in step 186 turns the delay off by making the level of the wet portion of the delay mix to zero.

When using the delay controller setup, notice that the numbers shown in the MRC's display range from 0-127. To turn these numbers into 602 numbers that match the front panel, you must convert them via the parameter tables found at the end of Appendix C. For instance, on the MRC, the feedback (**FB**) parameter varies from 0 to 127. If you set slider 3 on the MRC to minimum and then listen to the 602's output, the result should be intense echo with no signs of decay. Why is this? A glance at the Delay Feedback Table in Appendix C shows that sending the 602 a feedback value of 0 results in maximum negative (reverse polarity) feedback. To set the feedback to 0 (no feedback), you must set slider 3 on the MRC so the MRC's display reads 64.

C. MIDI Implementation Notes

This appendix describes the MIDI implementation of the 602. If you are a newcomer to MIDI, you would do well to familiarize yourself with MIDI and its usage by reading one of the many introductory-level books available at booksellers.

C.1 Overview

There are two MIDI messages of importance to the 602: MIDI Control Change and MIDI Sysex. The standard MIDI implementation table may be found in Appendix G.

The 602 responds to MIDI messages containing its unique device type and unit number as well as MIDI messages matching only its device type (provided that Omni mode has been turned on).

MIDI Control Change messages affect volume, panning, bank select, and omni mode. MIDI Program Change commands change user programs (in conjunction with the Bank Select command). All other 602 program changes (set/get program data, identify request) occur via MIDI sysex messages.

An identification scheme allows a daisy chain of 602's to share a MIDI bus. Sending an identification request to the first unit in the chain causes all units to report their current MIDI channel and unit number, along with the identifying string, "SYMETRIX 602". The responses are in the same order as the arrangement of units along the MIDI daisy chain.

In the following tables, all numbers are written in the base (decimal or hexadecimal) listed at the head of each table. Where necessary for clarity, hexadecimal numbers are followed with 'h', and decimal numbers are followed with 'd'). Type refers to the length of the request or response.

C.1.1 Control Change (B*n*)

The control change message commands all devices sharing a given MIDI channel to change one of the following parameters. Typically, only like devices share the same MIDI channel.

Code (dec)	Action	Value(dec)
7	Volume (input/output)	0-127 (-64dB - 0dB attn)
10	L/R Pan	0: l 64: c 127 r:
32	Bank Select	0: RAM (1-128) 1: ROM (129-256)
124	Omni Mode Off	0
125	Omni Mode On	0

C.1.1.1 Example

Command the 602 to set the output panning to center.

Send (hex): <MIDI command><nnnn><data>< data>
B*n* 0A 40
where: *n* is the MIDI channel number (0-F), encoded in hex

C.1.2 Realtime MIDI

There are two Realtime MIDI setups available per program. Each setup allows some predefined MIDI action to control any one parameter on the 602. An offset and scale factor (multiplier) tailor the response. These are stored on a per-program basis and can be edited through MIDI sysex or through the realtime MIDI functions on the front-panel (see Chapter 7). Regardless of the actual range of values required internally (to the 602), the externally accessed range of values is mapped across the range of 0 to 127 (decimal). The parameter value can be scaled and offset to shift the value into a useful range.

The range of the scale factor is plus or minus 4 and the Realtime Scaling Table maps the 0 to 127 range used in the edit buffer to the stored scale factor value. The Realtime Scaling Table may be found later in this chapter.

For instance, to represent a scale factor (multiplier value) of 2.4, refer to the Realtime Scaling Table, locate the value 2.4 within the table grid, and read the step number from the row and

column headings (119 in this example). From MIDI, you send an edit buffer data set command (1Ch) with a parameter offset value of 4Ah and a parameter value of 77h (119d).

The offset value is added after multiplication by the scale factor and is stored with its own offset of -64. The stored offset is doubled before being applied to the realtime MIDI value. The offset is stored in the 602 as an unsigned value having a range of 0-127. Each step in the stored value between 0 and 127 represents an offset increment of 2, and the actual offset is derived from the stored offset as follows (dec):

$offset = (v - 64) * 2$ where: v = stored value (0-127) and *offset* = (-128 to +127)

The value applied to the edit buffer is derived as follows:

$e = (m * k) + v$ where: e = new edit buffer value,
m = modulation value
v = stored offset,
k = stored scaling value from table

Note: in the 602, the scale factor is shown as **SCAL** on the display.

For the purposes of Realtime MIDI, each edit buffer parameter has a range of 0-127 regardless of what the actual range of values is, as specified in the tables at the end of this appendix. Thus an on/off type of parameter will be **off** if the result of the offset and scale operation ranges from 0 to 63 and **on** if the result is in the range of 64-127.

Realtime Block 1 has two additional parameters that apply an upper (CLPH/clip hi) and lower (CLPL/clip lo) limit to the final parameter value. You can use these parameters to keep a realtime MIDI value within a useful range. Realtime Block 2 has an additional modulation source, **bLC1** that is the output of Realtime Block 1.

As each parameter change request arrives, it immediately modifies the appropriate edit buffer location, and inserts a DSP update request into a 128 byte FIFO queue. The 602 processes the queue as time allows. If a burst of requests fills the queue, new requests are discarded until there is room in the queue. If your MIDI controller spews data and overruns the 602's queue, the 602 may ignore the extra data. If the stream of data ends before the 602 finishes processing its queue, the 602 may miss the last message in the data stream. A program change forcefully clears the queue.

C.1.3 Sysex Implementation (F0)

All sysex messages use the universal system exclusive code format. The MIDI sysex message uses the following format (hex):

Send: <sysex><mfrID><unitID><unit#><command><data ...><EOX>
<F0><00><00><5E><02><#><command><data ...><F7>

Send a Edit Buffer Data Set message to the 602, and set the level for Filter 1 of the parametric equalizer block to +12 dB:

<sysex><mfrID><unitID><unit#><command><offset><value><EOX>
<F0><00><00><5E><02><#><1C><06><70><F7>

where: F0 is the midi sysex command
00 00 5E is Symetrix' ID
02 identifies the 602
is the unit number
1C is the Edit Buffer Data Set command (see Table 3)
06 is the level parameter from the Filter 1 table (Table 11)
70 is the value for +12 dB from Table 27 (see also Table 11)

In the tables that follow:

- Short data transfers are from or to the edit buffer only.
- Block data transfers can access any of the stored program data including the edit buffer and system setup.
- All offsets are in decimal.
- (INPUT)/(OUTPUT) refer to the 602.
- REQUEST is a data request to the 602
- RESPONSE is data from the 602.

C.1.4 Sysex Echo

There are several conditions under which sysex messages are echoed through the 602 to the MIDI OUT connector:

- The message's manufacturer's ID or product identifier is for a different product.
- The message's destination number does not match.
- The message's command is not recognized.
- The message's destination unit number does not match.
- The message's destination unit number is the 'omni' value of 7fh.
- The 602 recognizes the message but is in unit 'ALL' mode.

If the 602 recognizes the sysex message (and the message was specifically addressed to the particular 602), then the message is absorbed and not echoed.

C.1.5 Recognized MIDI Commands

The 602 recognizes the following MIDI sysex messages:

Cmd# (hex)	Command Description
11	Edit buffer data request
1	Edit buffer data response
1C	Edit buffer data set
12	Program/Setup data request
1D	Data response
1D	Program/Setup data write
13	Identity Request
3	Identity Response

Table 1. Edit Buffer Data Request

REQUEST TO 602		
Type Short	**Requests** One parameter by number	
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfdID 1>	0
3	<mfdID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F(all)
6	<command>	11
7	<parameter offset>	0-7F
8	<EOX>	F7

Table 2. Edit Buffer Data Response

DATA FROM 602		
Type Short	**Returns** One parameter in Edit Buffer	
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfdID 1>	0
3	<mfdID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	1
7	<parameter #>	0-7F
8	<parameter value>	0-7F
9	<EOX>	F7

Table 3. Edit Buffer Data Set

DATA TO 602		
Type Short	**Returns** One parameter in Edit Buffer	
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	1C
7	<parameter offset >	0-7F
8	<parameter value>	0-7F
9	<EOX>	F7

Table 4. Program/Setup Data Request

REQUEST TO 602		
Type Long	**Requests** Block of parameters by address	
See Parameter Map for address Range (hex)		
Off (dec)	**Value (hex)**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	12
7	<Offset to start of data, top 2 bits>	
8	<Offset, middle 7 bits>	
9	<Offset, bottom 7 bits>	
10	<number of bytes, top 2 bits>	special case: 0,0,0 - transmit one edit buffer size worth
11	<number of bytes, middle 7 bits>	special case: 7F, 7F, 7F - transmit all
12	<number, bottom 7 bits>	
13	<EOX>	F7

Table 5. Data Response

RESPONSE FROM 602		
Type Long	**Requests** Block of parameters by Offset (dec)	
Data ordering Edit Buffer | 128 User Programs | 128 ROM Programs | Machine Setup		
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	0
5	<unit/channel>	0-7E, 7F (all)
6	<command>	1D
7	<Offset to start of data, top 2 bits>	
8	<Offset, middle 7 bits>	
9	<Offset, bottom 7 bits>	
10	<number of bytes, top 2 bits>	
11	<number, middle 7 bits>	
12	<number, bottom 7 bits>	
13	<first parameter>	0-7F
-	...	
-	...	
13+bytes	<EOX>	F7

Table 6. Program/Setup Data Write

DATA TO 602		
Type Long	**Purpose** Write a block of parameters by address	
Data ordering Edit Buffer | 128 User Programs | 128 ROM Programs | Machine Setup		
Off. (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	1D
7	<Offset to start of data, top 2 bits>	
8	<Offset to start of data, middle 7 bits>	
9	<Offset, bottom 7 bits>	
10	<number of bytes, top 2 bits>	
11	<number, middle 7 bits>	
12	<number, bottom 7 bits>	
13	data bytes	
...	data	
...	data	
	<EOX>	F7

Table 7. Identify Request

REQUEST TO 602		
	Purpose Request the identity of a 602.	
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	13
7	<EOX>	F7

Table 8. Identify Response

RESPONSE FROM 602		
	Purpose 602 response to Identify Request (13h).	
Off (dec)	**Value**	**Range (hex)**
0	<sysex>	F0
1	<mfrID 0>	0
2	<mfrID 1>	0
3	<mfrID 2>	5E
4	<device type>	2
5	<unit/channel>	0-7E, 7F (all)
6	<command>	3
7	<MIDI channel#>	0-0F, 7F (omni)
8	'S'	
9	'Y'	
10	'M'	
11	'E'	
12	'T'	
13	'R'	
14	'I'	
15	'X'	
16	' '	
17	'6'	
18	'0'	
19	'2'	
20	<EOX>	F7

The parameter map shows the location of various entities within the memory space of the 602. You can access these by using the program/setup data write command (1Dh).

Table 9. Parameter Map

Offset (dec)	Description	
0-99	Edit Buffer	
100-199	RAM Program 1	
12800-12899	RAM Program 128	
12900-12999	ROM Program 1	
25600-25699	ROM Program 128	
25700-25835	Global Parameters	
Global Parameter Offset (dec)	**Description**	**Value (dec)**
25700	MIDI Channel	0-15 127:OMNI
25701	MIDI Unit	0-126 127: all
25702	Current Pgm	range: 0-127 LSB 7 bits program number, 0-255
25703	Current Pgm	0-1 MSB 2 bits of program number
25704	Reserved	
25705	Signal Source	0: AES input 1 to L & R 1: AES input 2 to L & R 2: AES inputs summed to L & R 3: AES inputs to L & R (stereo) 4: Analog input 1 to L & R 5: Analog input 2 to L & R 6: Analog inputs summed to L & R 7:Analog inputs to L & R, independent gain controls (stereo) 8:Analog inputs to L & R, ganged gain controls (stereo)
25706	Signal/Clock Configuration	BIT0: 0:DSP->DAC 1:AES->DAC BIT1: 0:ADC->MCLK 1:AES->MCLK BIT2: 0:ADC->INPUT 1:AES->INPUT BIT4: 0:44.1 kHz sample rt 1:48.0 kHz sample rt
25707	MIDI echo	0: no echo 1: echo
25708	Memory protect	0: no protection 1: protected
25709	Reserved	
25710	Reserved	
25711	Front panel lockout	0:enabled, 85: partial 127:maximum
25712	Reserved	
25713	Current sample rate	1:48khz, 2:44.1khz (read only)
25714-5	full left pan	see below
25774-5	center	
25834-5	full right pan	

The pan table (locations 25714-25835) is loaded from the default pan table when the 602 is first initialized . The table consists of 61 pairs of left-right attenuation values. These values may be edited should your application require something different. The new values survive power on/off cycles.

C.1.6 Data Structure Per Program

All programs use the following data structure. Each program parameter has a specified offset within the Edit Buffer. By reading or writing these parameters, you can query or set particular program parameters. By dumping the entire range (0-99d), you can look at the status of the entire edit buffer. By reloading the same range, you can superimpose your own values onto the same parameters. (You can also modifiy any value individually via MIDI.) You'll need an external MIDI program editor to perform this task. For clarity, each function is presented in its own table, however the offsets shown refer to a contiguous block of memory in the unit.

Table 10. Global

Offset	Description	Range (dec)	Reference
0	Dynamics sidechain filter mode	0: highpass shelving 127: lowpass	See also offset 65
1	De-esser sidechain filter frequency	0: default, 5 kHz 1:127	See Frequency Table
2	Input Gain		See Attn18 Table

Table 11. Filter 1

Offset (dec)	Description	Range (dec)	Reference
3	F1 Mode	64: shelving 127: bp	
4	F1 Freq		See Frequency Table
5	F1 Q		See BW Table
6	F1 Level	81: out	See Attn82 Table
7	F1 Freq/BW Rate of Change		See Tc Table
8	F1 Level Rate of Change		See Tc Table

Table 12. Filter 2

Offset		Description	Range (dec)	Reference
dec	hex		dec	
9	9	F2 Mode	(NOP, always BP)	
10	A	F2 Freq		See Frequency Table
11	B	F2 BW		See BW Table
12	C	F2 Level	81: out	See Attn82 Table
13	D	F2 Freq/BW Rate of Change		See Tc Table
14	E	F2 Level Rate of Change		See Tc Table

Table 13. Filter 3

Offset		Description	Range	Reference
dec	hex		dec	
15	F	F3 Mode	64: shelving 127: bp	
16	10	F3 Freq		See Frequency Table
17	11	F3 BW		See BW Table
18	12	F3 Level	81: out	See Attn82 Table
19	13	F3 Freq/BW Rate of Change		See Tc Table
20	14	F3 Level Rate of Change		See Tc Table

Table 14. De-ess Processor				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
21	15	Mode	0: off 127: on	
22	16	Absolute Threshold		See Attn100 Table
23	17	Relative Threshold		See Attn100 Table
24	18	Attack Time		See Tc Table
25	19	Release Time		See Tc Table

Table 15. Noise Reduction Processor				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
26	1A	Mode	0: off 127: on	
27	1B	Minimum Turnover Freq		See Frequency Table
28	1C	Reserved		.
29	1D	Absolute Threshold		See Attn100 Table
30	1E	Relative Threshold		See Attn100 Table
31	1F	Reserved		
32	20	Reserved		

Table 16. Delay Processor				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
33	21	Mode	0: off 127: on	
34	22	Delay Line 1 Delay Time	0-127 0: 500 µs 127: 330 ms	See DelayTime Table
35	23	Delay Line 2 Delay Time	0-127 0: 500µS 127: 330ms	See DelayTime Table
36	24	Delay Line Rate of Change	0-127	See Tc Table
37	25	Cross Recirculation Attenuation	0: pos 0 dB 64: off 127: neg 0 dB	See Delay Feedback Table
38	26	Filter Frequency	0-127	See Delay Filter Table
39	27	Direct/Delay Mix Percent	0: 0% delay, 100% direct 127: 100% delay, 0% direct	See Normalized MIDI Pan Input Table CH 1 pan tbl: Direct CH 2 pan tbl: Delay
40	28	Modulation Depth	0-127	
41	29	Modulation Rate	=(value x .1 Hz + .1 Hz)	
42	2A	Modulation Type	0: random 85: sine 127: triangle	
43	2B	Mix Rate of Change		See Tc Table

Table 17. Expansion Parameters				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
44	2C	Mode	0: off 127: on	
45	2D	Threshold		See Attn100 Table
46	2E	Attack Time		See Tc Table
47	2F	Release Time		See Tc Table
48	30	Expansion Ratio		See Expansion Ratio Table
49	31	Knee Control	0: 6dB 43: 12 dB 85: 18 dB 127: 24 dB	

Table 18. Compression Parameters				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
50	32	Compressor Mode	0: out 64: AGC 127: compressor	
51	33	Threshold		See Attn100 Table
52	34	Attack Time		See Tc Table
53	35	Release Time		See Tc Table
54	36	Compression Ratio		See Compression Ratio Table
55	37	Knee Control	0: 6dB 43: 12 dB 85: 18 dB 127: 24 dB	
83	53	Makeup Gain	0-127 0: auto 1: 0 dB makeup gain 127:24 dB makeup gain	Shared with AGC See Attn24 Table

Table 19. AGC Parameters				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
56	38	Threshold		See Attn100 Table
57	39	Attack Time		See Tc Table
58	3A	Release Time		See Tc Table
59	3B	Compression Ratio		See Compression Ratio Table
60	3C	Knee Control	0: 6dB 43: 12 dB 85: 18 dB 127: 24 dB	
83	53	Makeup Gain	0-127 0: auto 1: 0 dB makeup gain 127:24 dB makeup gain	Shared with compressor See Attn24 Table

Table 20. ARM Sense Parameters				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
61	3D	Auto Release Threshold		See Attn100 Table
62	3E	ARM Peak Release Tc		See Tc Table
63	3F	ARM Integration Tc		See Tc Table
64	40	ARM Threshold		See ARM Threshold Table

Table 21. LOG Converter Parameters				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
65	41	Control Chain Turnover Frequency	0-127	See Frequency Table See also offset 0
66	42	Log Averaging Filter Tc		See Tc Table
67	43	Sidechain Lookahead	0: 0 us 127: 2.6 ms @ 48 kHz linear scale	See Sidechain Lookahead Table

Table 22. Output				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
68	44	Output Attenuation		See Output Level Table
69	45	Output Pan	0-127 0: ch 1 max 64: center 127: ch 2 max	See Default Pan Table
70	46	Output Gain Rate of Change		See Tc Table

Table 23. Realtime MIDI Block 1				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
71	47	Control Type	0: None/Off/parameter edit 1: Control Change 2: Aftertouch 3: Pitch bend (msb 7 bit) 4: Delay section modulation oscillator 1 5: Delay section modulation oscillator 2 6: Log signal level, dynamics section 7: NR center freq 8: Instantaneous gain-reduction value, compressor 9: Instantaneous gain-reduction value, expander	
72	48	3 Byte MIDI Message, Second Parameter	0-127	
73	49	Control Offset (dec)	0-127 64: 0 0: -128 127:+127	
74	4A	Control Scaling	0-127 64: No Effect 0: -4 127: +4	Realtime Scaling Table
75	4B	Parameter to Modify (Offset (dec))	Edit Buffer Offset Address (0-70)	
81	51	Realtime Block 1 Floor Clip	0-127	Minimum normalized edit buffer value
82	52	Realtime Block 1 Ceiling Clip	0-127 0: 127 127: 0	Maximum normalized edit buffer value.

Table 24. Realtime MIDI Block 2				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
76	4C	Control Type	0: None/Off (default) 1: Control Change 2: Aftertouch 3: Pitch bend (msb 7 bit) 4: Delay section modulation oscillator 1 5: Delay section modulation oscillator 2 6: Log signal level, dynamics section 7: NR center freq 8: Instantaneous gain-reduction value, compressor 9: Instantaneous gain-reduction value, expander 10: Block 1 output	
77	4D	3 Byte MIDI Message Second Parameter	0-127	
78	4E	Control Offset (dec)	0-127 64: 0 0: -128 127:+127	
79	4F	Control Scaling	0-127 64: No Effect 0: -4 127: +4	Realtime Scaling Table
80	50	Parameter to Modify (Offset (dec))	Edit Buffer Offset Address (0-70)	

The Program Name (offset 84-99) is not accessible from the front panel; it is only accessible via MIDI. You can use this with an external MIDI editor to give your 602 programs meaningful (to you) names.

Table 25. Miscellaneous				
Offset		**Description**	**Range**	**Reference**
dec	hex		dec	
81	51	Realtime Block 1 Floor Clip	0-127	Block 1 only. Minimum normalized edit buffer value
82	52	Realtime Block 1 Ceiling Clip	0-127 0: 127 127: 0	Block 1 only. Maximum normalized edit buffer value.
83	53	Makeup Gain	0-127 0: auto makeup gain 1: 0 dB makeup gain 127: 24 dB makeup gain	Shared between AGC & Compressor Attn24 Table
84-99	54-63	Program name		ASCII program name
99	63	Modified flag		Forced to "*" if program modified from front panel

C.1.7 MIDI Parameter Tables

Many of the parameters used in the 602 are extracted from tables. When controlling the 602 via MIDI, all values sent to the 602 via its MIDI port must be mapped from their real-world values into a table based on 128-steps. The following tables list various system parameters and their conversion values.

Table 26. Attn18 Table (dB)

	0	1	2	3	4	5	6	7	8	9
0:	-18.0	-18.0	-18.0	-17.0	-17.0	-16.0	-16.0	-15.0	-15.0	-14.0
10:	-14.0	-13.0	-13.0	-12.0	-12.0	-11.0	-11.0	-10.0	-10.0	-9.0
20:	-9.0	-8.0	-8.0	-8.0	-7.0	-7.0	-6.5	-6.5	-6.0	-6.0
30:	-5.5	-5.5	-5.0	-5.0	-4.5	-4.5	-4.0	-4.0	-3.5	-3.5
40:	-3.0	-3.0	-2.5	-2.5	-2.5	-2.0	-2.0	-1.5	-1.5	-1.0
50:	-1.0	-0.5	-0.5	+0.0	+0.0	+0.5	+0.5	+1.0	+1.0	+1.5
60:	+1.5	+2.0	+2.0	+3.0	+3.0	+3.0	+3.5	+3.5	+4.0	+4.0
70:	+4.5	+4.5	+5.0	+5.0	+5.5	+5.5	+6.0	+6.0	+6.5	+6.5
80:	+7.0	+7.0	+7.5	+7.5	+8.0	+8.0	+8.0	+8.5	+8.5	+9.0
90:	+9.0	+9.5	+9.5	+10.0	+10.0	+10.5	+10.5	+11.0	+11.0	+11.5
100:	+11.5	+12.0	+12.0	+12.5	+12.5	+13.0	+13.0	+13.0	+13.5	+13.5
110:	+14.0	+14.0	+14.5	+14.5	+15.0	+15.0	+15.5	+15.5	+16.0	+16.0
120:	+16.5	+16.5	+17.0	+17.0	+17.5	+17.5	+18.0	+18.0		

Table 27. Attn82 Table (dB)

	0	1	2	3	4	5	6	7	8	9
0:	-82.0	-82.0	-80.0	-78.0	-76.0	-76.0	-74.0	-72.0	-70.0	-68.0
10:	-68.0	-66.0	-64.0	-62.0	-62.0	-60.0	-58.0	-56.0	-54.0	-54.0
20:	-52.0	-50.0	-48.0	-48.0	-46.0	-44.0	-42.0	-40.0	-40.0	-38.0
30:	-36.0	-34.0	-32.0	-32.0	-31.0	-30.0	-29.0	-29.0	-28.0	-27.0
40:	-26.0	-25.0	-25.0	-24.0	-23.0	-22.0	-22.0	-21.0	-20.0	-19.0
50:	-18.0	-18.0	-17.0	-16.0	-15.0	-15.0	-14.0	-13.0	-12.0	-11.0
60:	-11.0	-10.0	-9.0	-8.0	-7.0	-7.0	-6.0	-5.5	-5.0	-5.0
70:	-4.5	-4.0	-3.5	-3.0	-3.0	-2.5	-2.0	-1.5	-1.5	-1.0
80:	-0.5	+0.0	+0.5	+0.5	+1.0	+1.5	+2.0	+2.0	+2.5	+3.0
90:	+3.5	+4.0	+4.0	+4.5	+5.0	+5.5	+6.0	+6.0	+6.5	+7.0
100:	+7.5	+7.5	+8.0	+8.5	+9.0	+9.5	+9.5	+10.0	+10.5	+11.0
110:	+11.0	+11.5	+12.0	+12.5	+13.0	+13.0	+13.5	+14.0	+14.5	+14.5
120:	+15.0	+15.5	+16.0	+16.5	+16.5	+17.0	+17.5	+18.0		

Table 28. Attn100 Table (dB)

	0	1	2	3	4	5	6	7	8	9
0:	-100	-100	-98.0	-96.0	-94.0	-94.0	-92.0	-90.0	-88.0	-86.0
10:	-86.0	-84.0	-82.0	-80.0	-80.0	-78.0	-76.0	-74.0	-72.0	-72.0
20:	-70.0	-68.0	-66.0	-66.0	-64.0	-62.0	-60.0	-58.0	-58.0	-56.0
30:	-54.0	-52.0	-50.0	-50.0	-49.0	-48.0	-47.0	-47.0	-46.0	-45.0
40:	-44.0	-43.0	-43.0	-42.0	-41.0	-40.0	-40.0	-39.0	-38.0	-37.0
50:	-36.0	-36.0	-35.0	-34.0	-33.0	-33.0	-32.0	-31.0	-30.0	-29.0
60:	-29.0	-28.0	-27.0	-26.0	-25.0	-25.0	-24.0	-23.5	-23.0	-23.0
70:	-22.5	-22.0	-21.5	-21.0	-21.0	-20.5	-20.0	-19.5	-19.5	-19.0
80:	-18.5	-18.0	-17.5	-17.5	-17.0	-16.5	-16.0	-16.0	-15.5	-15.0
90:	-14.5	-14.0	-14.0	-13.5	-13.0	-12.5	-12.0	-12.0	-11.5	-11.0
100:	-10.5	-10.5	-10.0	-9.5	-9.0	-8.5	-8.5	-8.0	-7.5	-7.0
110:	-7.0	-6.5	-6.0	-5.5	-5.0	-5.0	-4.5	-4.0	-3.5	-3.5
120:	-3.0	-2.5	-2.0	-1.5	-1.5	-1.0	-0.5	0.0		

Table 29. Parametric Bandwidth Table (in octaves)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	0.050	0.050	0.050	0.050	0.055	0.055	0.055	0.060	0.060	0.060
10:	0.065	0.065	0.065	0.070	0.070	0.070	0.075	0.075	0.075	0.075
20:	0.080	0.080	0.080	0.085	0.085	0.085	0.090	0.090	0.090	0.095
30:	0.095	0.095	0.10	0.10	0.10	0.10	0.20	0.20	0.20	0.30
40:	0.30	0.30	0.40	0.40	0.40	0.50	0.50	0.50	0.60	0.60
50:	0.60	0.60	0.70	0.70	0.70	0.80	0.80	0.80	0.90	0.90
60:	0.90	1.0	1.0	1.0	1.1	1.1	1.1	1.1	1.2	1.2
70:	1.2	1.3	1.3	1.3	1.4	1.4	1.4	1.5	1.5	1.5
80:	1.6	1.6	1.6	1.6	1.7	1.7	1.7	1.8	1.8	1.8
90:	1.9	1.9	1.9	2.0	2.0	2.0	2.1	2.1	2.1	2.1
100:	2.2	2.2	2.2	2.3	2.3	2.3	2.4	2.4	2.4	2.5
110:	2.5	2.5	2.6	2.6	2.6	2.6	2.7	2.7	2.7	2.8
120:	2.8	2.8	2.9	2.9	2.9	3.0	3.0	3.0		

Table 30. Frequency Table (Hz)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	31	31	33	36	36	38	41	44	44	47
10:	50	54	54	58	60	67	67	72	77	82
20:	82	88	95	100	100	109	120	125	125	134
30:	144	154	154	165	177	177	189	203	218	218
40:	233	250	268	268	287	308	330	330	354	379
50:	406	406	435	467	500	500	536	574	616	616
60:	660	707	758	758	812	871	871	933	1000	1072
70:	1072	1149	1231	1320	1320	1414	1516	1625	1625	1741
80:	1866	2000	2000	2144	2297	2462	2462	2639	2828	3031
90:	3031	3249	3482	3732	3732	4000	4287	4287	4595	4925
100:	5278	5278	5657	6063	6498	6498	6964	7464	8000	8000
110:	8574	9190	9849	9849	10556	11314	12126	12126	12996	13929
120:	14929	14929	16000	17148	18379	18379	19698	21112		

Table 31. Output Level Table (dB)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	OFF	OFF	-90.0	-88.0	-84.0	-82.0	-82.0	-80.0	-78.0	-76.0
10:	-74.0	-72.0	-72.0	-70.0	-68.0	-66.0	-64.0	-62.0	-62.0	-60.0
20:	-58.0	-56.0	-54.0	-52.0	-52.0	-50.0	-49.0	-48.0	-47.0	-46.0
30:	-46.0	-45.0	-44.0	-43.0	-42.0	-42.0	-41.0	-40.0	-39.0	-38.0
40:	-37.0	-37.0	-36.0	-35.0	-34.0	-33.0	-32.0	-32.0	-31.0	-30.0
50:	-29.0	-28.0	-27.0	-27.0	-26.0	-25.0	-24.0	-23.0	-22.0	-22.0
60:	-21.0	-20.0	-19.0	-18.0	-17.0	-17.0	-16.0	-15.0	-14.0	-13.0
70:	-13.0	-12.0	-11.0	-10.0	-9.5	-9.0	-9.0	-8.5	-8.0	-7.5
80:	-7.0	-6.5	-6.5	-6.0	-5.5	-5.0	-4.5	-4.0	-4.0	-3.5
90:	-3.0	-2.5	-2.0	-1.5	-1.5	-1.0	-0.5	+0.0	+0.5	+0.5
100:	+1.0	+1.5	+2.0	+2.5	+3.0	+3.0	+3.5	+4.0	+4.5	+5.0
110:	+5.5	+5.5	+6.0	+6.5	+7.0	+7.5	+8.0	+8.0	+8.5	+9.0
120:	+9.5	+10.0	+11.0	+11.0	+12.0	+13.0	+14.0	+15.0		

Table 32. Expander Ratio Table										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0
10:	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2
20:	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5
30:	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8
40:	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0
50:	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.5
60:	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.5
70:	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	4.0
80:	4.0	4.0	4.0	4.0	4.0	4.0	4.0	4.0	4.0	5.0
90:	5.0	5.0	5.0	5.0	5.0	5.0	5.0	5.0	5.0	6.0
100:	6.0	6.0	6.0	6.0	6.0	6.0	6.0	6.0	6.0	7.0
110:	7.0	7.0	7.0	7.0	7.0	7.0	7.0	7.0	7.0	8.0
120:	8.0	8.0	8.0	8.0	8.0	8.0	8.0	8.0		

Table 33. Compressor Ratio Table										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.2
10:	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.5	1.5
20:	1.5	1.5	1.5	1.5	1.5	1.5	1.8	1.8	1.8	1.8
30:	1.8	1.8	1.8	1.8	1.8	2.0	2.0	2.0	2.0	2.0
40:	2.0	2.0	2.0	2.5	2.5	2.5	2.5	2.5	2.5	2.5
50:	2.5	2.5	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0
60:	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	4.0
70:	4.0	4.0	4.0	4.0	4.0	4.0	4.0	5.0	5.0	5.0
80:	5.0	5.0	5.0	5.0	5.0	5.0	6.0	6.0	6.0	6.0
90:	6.0	6.0	6.0	6.0	7.0	7.0	7.0	7.0	7.0	7.0
100:	7.0	7.0	7.0	8.0	8.0	8.0	8.0	8.0	8.0	8.0
110:	8.0	9.0	9.0	9.0	9.0	9.0	9.0	9.0	9.0	9.0
120:	10.0	10.0	10.0	10.0	10.0	10.0	10.0	10.0		

Table 34. AGC Ratio Table										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0	1.0
10:	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2	1.2
20:	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5	1.5
30:	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8	1.8
40:	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0	2.0
50:	2.2	2.2	2.2	2.2	2.2	2.2	2.2	2.2	2.2	2.2
60:	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.5	2.8
70:	2.8	2.8	2.8	2.8	2.8	2.8	2.8	2.8	2.8	3.0
80:	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.0	3.2
90:	3.2	3.2	3.2	3.2	3.2	3.2	3.2	3.2	3.2	3.5
100:	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.5	3.8
110:	3.8	3.8	3.8	3.8	3.8	3.8	3.8	3.8	3.8	4.0
120:	4.0	4.0	4.0	4.0	4.0	4.0	4.0	4.0		

Table 35. ARM Threshold (dB)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	OFF	OFF	OFF	OFF	-0.1	-0.1	-0.1	-0.1	-0.2	-0.2
10:	-0.2	-0.2	-0.4	-0.4	-0.4	-0.4	-0.6	-0.6	-0.6	-0.6
20:	-0.8	-0.8	-0.8	-0.8	-1.0	-1.0	-1.0	-1.0	-1.2	-1.2
30:	-1.2	-1.2	-1.5	-1.5	-1.5	-1.5	-1.8	-1.8	-1.8	-1.8
40:	-2.0	-2.0	-2.0	-2.0	-2.2	-2.2	-2.2	-2.2	-2.5	-2.5
50:	-2.5	-2.5	-2.8	-2.8	-2.8	-2.8	-3.0	-3.0	-3.0	-3.0
60:	-3.2	-3.2	-3.2	-3.2	-3.5	-3.5	-3.5	-3.5	-3.8	-3.8
70:	-3.8	-3.8	-4.0	-4.0	-4.0	-4.0	-4.5	-4.5	-4.5	-4.5
80:	-5.0	-5.0	-5.0	-5.0	-5.5	-5.5	-5.5	-5.5	-6.0	-6.0
90:	-6.0	-6.0	-6.5	-6.5	-6.5	-6.5	-7.0	-7.0	-7.0	-7.0
100:	-7.5	-7.5	-7.5	-7.5	-8.0	-8.0	-8.0	-8.0	-8.5	-8.5
110:	-8.5	-8.5	-9.0	-9.0	-9.0	-9.0	-9.5	-9.5	-9.5	-9.5
120:	-10.0	-10.0	-10.0	-10.0	-12.0	-12.0	-12.0	-12.0		

Table 36. Time Constant Table										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	100µs	150µs	200µs	250µs	300µs	350µs	400µs	500µs	600µs	700µs
10:	800µs	900µs	1.0ms	1.5ms	2.0ms	2.5ms	3.0ms	3.5ms	4.0ms	4.5ms
20:	5.0ms	5.5ms	6.0ms	7.0ms	8.0ms	9.0ms	10ms	12ms	14ms	16ms
30:	18ms	20ms	22ms	24ms	26ms	28ms	30ms	32ms	34ms	36ms
40:	38ms	40ms	45ms	50ms	55ms	60ms	65ms	70ms	75ms	80ms
50:	85ms	90ms	95ms	100ms	110ms	120ms	130ms	140ms	150ms	160ms
60:	170ms	180ms	190ms	200ms	210ms	220ms	230ms	240ms	250ms	260ms
70:	270ms	280ms	290ms	300ms	320ms	340ms	360ms	380ms	400ms	420ms
80:	440ms	460ms	480ms	500ms	520ms	540ms	560ms	580ms	600ms	620ms
90:	640ms	660ms	680ms	700ms	725ms	750ms	775ms	800ms	850ms	900ms
100:	925ms	950ms	975ms	1.0s	1.1s	1.2s	1.3s	1.4s	1.5s	1.6s
110:	1.7s	1.8s	1.9s	2.0s	2.5s	3.0s	3.5s	4.0s	4.5s	5.0s
120:	5.5s	6.0s	6.5s	7.0s	7.5s	8.0s	9.0s	10.0s		

Table 37. Compresser/Expander Knee Table (in db)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	6	6	6	6	6	6	6	6	6	6
10:	6	6	6	6	6	6	6	6	6	6
20:	6	6	6	6	6	6	6	6	6	6
30:	6	6	12	12	12	12	12	12	12	12
40:	12	12	12	12	12	12	12	12	12	12
50:	12	12	12	12	12	12	12	12	12	12
60:	12	12	12	12	18	18	18	18	18	18
70:	18	18	18	18	18	18	18	18	18	18
80:	18	18	18	18	18	18	18	18	18	18
90:	18	18	18	18	18	18	24	24	24	24
100:	24	24	24	24	24	24	24	24	24	24
110:	24	24	24	24	24	24	24	24	24	24
120:	24	24	24	24	24	24	24	24		

Table 38. Makeup Gain Table (Attn24)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	+0.0	+0.0	+0.0	+0.5	+0.5	+0.5	+1.0	+1.0	+1.5	+1.5
10:	+1.5	+2.0	+2.0	+2.0	+2.5	+2.5	+3.0	+3.0	+3.0	+3.5
20:	+3.5	+4.0	+4.0	+4.0	+4.5	+4.5	+4.5	+5.0	+5.0	+5.5
30:	+5.5	+5.5	+6.0	+6.0	+6.5	+6.5	+6.5	+7.0	+7.0	+7.0
40:	+7.5	+7.5	+8.0	+8.0	+8.0	+8.5	+8.5	+8.5	+9.0	+9.0
50:	+9.5	+9.5	+9.5	+10.0	+10.0	+10.5	+10.5	+10.5	+11.0	+11.0
60:	+11.0	+11.5	+11.5	+12.0	+12.0	+12.0	+12.5	+12.5	+13.0	+13.0
70:	+13.0	+13.5	+13.5	+13.5	+14.0	+14.0	+14.5	+14.5	+14.5	+15.0
80:	+15.0	+15.5	+15.5	+15.5	+16.0	+16.0	+16.0	+16.5	+16.5	+17.0
90:	+17.0	+17.0	+17.5	+17.5	+17.5	+18.0	+18.0	+18.5	+18.5	+18.5
100:	+19.0	+19.0	+19.5	+19.5	+19.5	+20.0	+20.0	+20.0	+20.5	+20.5
110:	+21.0	+21.0	+21.0	+21.5	+21.5	+22.0	+22.0	+22.0	+22.5	+22.5
120:	+22.5	+23.0	+23.0	+23.5	+23.5	+23.5	+24.0	+24.0		

Table 39. Sidechain Lookahead Time (ms) Table										
0	1	2	3	4	5	6	7	8	9	
0:	0.00	0.02	0.04	0.06	0.08	0.10	0.12	0.15	0.17	0.19
10:	0.21	0.23	0.25	0.27	0.29	0.31	0.33	0.35	0.37	0.40
20:	0.42	0.44	0.46	0.48	0.50	0.52	0.54	0.56	0.58	0.60
30:	0.62	0.65	0.67	0.69	0.71	0.73	0.75	0.77	0.79	0.81
40:	0.83	0.85	0.87	0.90	0.92	0.94	0.96	0.98	1.00	1.02
50:	1.04	1.06	1.08	1.10	1.12	1.15	1.17	1.19	1.21	1.23
60:	1.25	1.27	1.29	1.31	1.33	1.35	1.37	1.40	1.42	1.44
70:	1.46	1.48	1.50	1.52	1.54	1.56	1.58	1.60	1.62	1.65
80:	1.67	1.69	1.71	1.73	1.75	1.77	1.79	1.81	1.83	1.85
90:	1.87	1.90	1.92	1.94	1.96	1.98	2.00	2.02	2.04	2.06
100:	2.08	2.10	2.12	2.15	2.17	2.19	2.21	2.23	2.25	2.27
110:	2.29	2.31	2.33	2.35	2.37	2.40	2.42	2.44	2.46	2.48
120:	2.50	2.52	2.54	2.56	2.58	2.60	2.62	2.65		

Table 40. Delay Time Table (ms)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	0.5	1.0	1.5	2.0	2.5	3.0	3.5	4.0	4.5	5.0
10:	6	7	8	9	10	11	12	13	14	15
20:	16	17	18	19	20	21	22	23	24	25
30:	26	27	28	29	30	32	34	36	38	40
40:	42	44	46	48	50	52	54	56	58	60
50:	62	64	66	68	70	72	74	76	78	80
60:	82	84	86	88	90	92	94	96	98	100
70:	105	110	115	120	125	130	135	140	145	150
80:	155	160	165	170	175	180	185	190	195	200
90:	205	210	215	220	225	230	235	240	245	250
100:	255	260	265	270	275	280	285	290	295	230
110:	235	240	245	250	255	260	265	270	275	280
120:	285	290	295	300	305	310	320	330		

Table 41. Delay Feedback Table: Negative then Positive feedback (dB)										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	0.0	0.0	-0.1	-0.2	-0.3	-0.4	-0.5	-1.0	-1.5	-1.5
10:	-2.0	-2.5	-3.0	-3.5	-4.0	-4.5	-5.0	-5.5	-5.5	-6.0
20:	-6.5	-7.0	-7.5	-8.0	-8.5	-9.0	-9.0	-9.5	-10.0	-11.0
30:	-12.0	-13.0	-14.0	-15.0	-16.0	-16.0	-17.0	-18.0	-19.0	-20.0
40:	-22.0	-24.0	-26.0	-26.0	-28.0	-30.0	-32.0	-34.0	-36.0	-38.0
50:	-40.0	-42.0	-42.0	-44.0	-46.0	-48.0	-50.0	-55.0	-60.0	-65.0
60:	-65.0	-70.0	-80.0	-90.0	-100	-90.0	-80.0	-70.0	-65.0	-65.0
70:	-60.0	-55.0	-50.0	-48.0	-46.0	-44.0	-42.0	-42.0	-40.0	-38.0
80:	-36.0	-34.0	-32.0	-30.0	-28.0	-26.0	-26.0	-24.0	-22.0	-20.0
90:	-19.0	-18.0	-17.0	-16.0	-16.0	-15.0	-14.0	-13.0	-12.0	-11.0
100:	-10.0	-9.5	-9.0	-9.0	-8.5	-8.0	-7.5	-7.0	-6.5	-6.0
110:	-5.5	-5.5	-5.0	-4.5	-4.0	-3.5	-3.0	-2.5	-2.0	-1.5
120:	-1.5	-1.0	-0.5	-0.4	-0.3	-0.2	-0.1	0.0		

Table 42. Delay Line Filter Table (Hz)										
0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	
0:	600	600	600	600	800	800	800	800	1000	1000
10:	1000	1000	1500	1500	1500	1500	2000	2000	2000	2000
20:	2500	2500	2500	2500	3000	3000	3000	3000	3500	3500
30:	3500	3500	4000	4000	4000	4000	4500	4500	4500	4500
40:	5000	5000	5000	5000	5500	5500	5500	5500	6000	6000
50:	6000	6000	6500	6500	6500	6500	7000	7000	7000	7000
60:	7500	7500	7500	7500	8000	8000	8000	8000	8500	8500
70:	8500	8500	9000	9000	9000	9000	9500	9500	9500	9500
80:	10000	10000	10000	10000	10500	10500	10500	10500	11000	11000
90:	11000	11000	11500	11500	11500	11500	12000	12000	12000	12000
100:	13000	13000	13000	13000	13500	13500	13500	13500	14000	14000
110:	14000	14000	15000	15000	15000	15000	16000	16000	16000	16000
120:	17000	17000	17000	17000	18000	18000	18000	18000		

Table 43. Realtime Scaling Table										
	0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
0:	-4.0	-3.8	-3.6	-3.4	-3.2	-3.0	-2.8	-2.6	-2.4	-2.2
10:	-2.0	-1.9	-1.8	-1.7	-1.6	-1.5	-1.4	-1.3	-1.2	-1.1
20:	-1.0	-0.95	-0.90	-0.85	-0.80	-0.75	-0.70	-0.65	-0.60	-0.58
30:	-0.56	-0.54	-0.52	-0.50	-0.48	-0.46	-0.44	-0.42	-0.40	-0.38
40:	-0.36	-0.34	-0.32	-0.30	-0.28	-0.26	-0.24	-0.22	-0.20	-0.18
50:	-0.16	-0.14	-0.12	-0.10	-0.09	-0.08	-0.07	-0.06	-0.05	-0.04
60:	-0.03	-0.02	-0.01	0.0	0.0	0.01	0.02	0.03	0.04	0.05
70:	0.06	0.07	0.08	0.09	0.10	0.12	0.14	0.16	0.18	0.20
80:	0.22	0.24	0.26	0.28	0.30	0.32	0.34	0.36	0.38	0.40
90:	0.42	0.44	0.46	0.48	0.50	0.52	0.54	0.56	0.58	0.60
100:	0.65	0.70	0.75	0.80	0.85	0.90	0.95	1.0	1.1	1.2
110:	1.3	1.4	1.5	1.6	1.7	1.8	1.9	2.0	2.2	2.4
120:	2.6	2.8	3.0	3.2	3.4	3.6	3.8	4.0		

Table 44. Default Pan Table					
Offset into Pan Table Buf	**Channel 1 Atten (dB)**	**Channel 2 Atten (dB)**	**Offset into Pan Table Buf**	**Channel 1 Atten (dB)**	**Channel 2 Atten (dB)**
0	-0.0	OFF	**30**	-3.0	-3.0
1	-0.0	-32.0	**31**	-3.5	-3.0
2	-0.0	-26.0	**32**	-3.5	-2.5
3	-0.0	-22.0	**33**	-4.0	-2.5
4	-0.0	-20.0	**34**	-4.0	-2.0
5	-0.0	-18.0	**35**	-4.5	-2.0
6	-0.0	-16.0	**36**	-5.0	-2.0
7	-0.0	-15.0	**37**	-5.0	-1.5
8	-0.0	-14.0	**38**	-5.5	-1.5
9	-0.0	-13.0	**39**	-5.5	-1.5
10	-0.0	-12.0	**40**	-6.0	-1.5
11	-0.0	-11.0	**41**	-6.5	-1.0
12	-0.5	-10.0	**42**	-7.0	-1.0
13	-0.5	-9.5	**43**	-7.5	-1.0
14	-0.5	-9.0	**44**	-8.0	-1.0
15	-0.5	-8.5	**45**	-8.5	-0.5
16	-1.0	-8.0	**46**	-9.0	-0.5
17	-1.0	-7.5	**47**	-9.5	-0.5
18	-1.0	-7.0	**48**	-10.0	-0.5
19	-1.0	-6.5	**49**	-11.0	-0.0
20	-1.5	-6.0	**50**	-12.0	-0.0
21	-1.5	-5.5	**51**	-13.0	-0.0
22	-1.5	-5.5	**52**	-14.0	-0.0
23	-1.5	-5.0	**53**	-15.0	-0.0
24	-2.0	-5.0	**54**	-16.0	-0.0
25	-2.0	-4.5	**55**	-18.0	-0.0
26	-2.0	-4.0	**56**	-20.0	-0.0
27	-2.5	-4.0	**57**	-22.0	-0.0
28	-2.5	-3.5	**58**	-26.0	-0.0
29	-3.0	-3.5	**59**	-32.0	-0.0
30	-3.0	-3.0	**60**	OFF	-0.0

Table 45. Normalized MIDI Pan Input Table											
MIDI Input Value	**Offset Into Pan Buffer**	**Chan. 1 Atten. (dB)**	**Chan. 2 Atten. (dB)**	**MIDI Input Value**	**Offset into Pan Buffer**	**Chan. 1 Atten. (dB)**	**Chan. 2 Atten (dB).**	**MIDI input Value**	**Offset Into Pan Buffer**	**Chan. 1 Atten. (dB)**	**Chan. 2 Atten. (dB)**
0	0	-0.0	OFF	**43**	20	-1.5	-6.0	**86**	41	-6.5	-1.0
1	0	-0.0	OFF	**44**	20	-1.5	-6.0	**87**	41	-6.5	-1.0
2	0	-0.0	OFF	**45**	21	-1.5	-5.5	**88**	41	-6.5	-1.0
3	1	-0.0	-32.0	**46**	21	-1.5	-5.5	**89**	42	-7.0	-1.0
4	1	-0.0	-32.0	**47**	22	-1.5	-5.5	**90**	42	-7.0	-1.0
5	2	-0.0	-26.0	**48**	22	-1.5	-5.5	**91**	43	-7.5	-1.0
6	2	-0.0	-26.0	**49**	23	-1.5	-5.0	**92**	43	-7.5	-1.0
7	3	-0.0	-22.0	**50**	23	-1.5	-5.0	**93**	44	-8.0	-1.0
8	3	-0.0	-22.0	**51**	24	-2.0	-5.0	**94**	44	-8.0	-1.0
9	4	-0.0	-20.0	**52**	24	-2.0	-5.0	**95**	45	-8.5	-0.5
10	4	-0.0	-20.0	**53**	25	-2.0	-4.5	**96**	45	-8.5	-0.5
11	5	-0.0	-18.0	**54**	25	-2.0	-4.5	**97**	46	-9.0	-0.5
12	5	-0.0	-18.0	**55**	26	-2.0	-4.0	**98**	46	-9.0	-0.5
13	6	-0.0	-16.0	**56**	26	-2.0	-4.0	**99**	47	-9.5	-0.5
14	6	-0.0	-16.0	**57**	27	-2.5	-4.0	**100**	47	-9.5	-0.5
15	7	-0.0	-15.0	**58**	27	-2.5	-4.0	**101**	48	-10.0	-0.5
16	7	-0.0	-15.0	**59**	28	-2.5	-3.5	**102**	48	-10.0	-0.5
17	8	-0.0	-14.0	**60**	28	-2.5	-3.5	**103**	49	-11.0	-0.0
18	8	-0.0	-14.0	**61**	29	-3.0	-3.5	**104**	49	-11.0	-0.0
19	9	-0.0	-13.0	**62**	29	-3.0	-3.5	**105**	50	-12.0	-0.0
20	9	-0.0	-13.0	**63**	30	-3.0	-3.0	**106**	50	-12.0	-0.0
21	10	-0.0	-12.0	**64**	30	-3.0	-3.0	**107**	50	-12.0	-0.0
22	10	-0.0	-12.0	**65**	30	-3.0	-3.0	**108**	51	-13.0	-0.0
23	10	-0.0	-12.0	**66**	31	-3.5	-3.0	**109**	51	-13.0	-0.0
24	11	-0.0	-11.0	**67**	31	-3.5	-3.0	**110**	52	-14.0	-0.0
25	11	-0.0	-11.0	**68**	32	-3.5	-2.5	**111**	52	-14.0	-0.0
26	12	-0.5	-10.0	**69**	32	-3.5	-2.5	**112**	53	-15.0	-0.0
27	12	-0.5	-10.0	**70**	33	-4.0	-2.5	**113**	53	-15.0	-0.0
28	13	-0.5	-9.5	**71**	33	-4.0	-2.5	**114**	54	-16.0	-0.0
29	13	-0.5	-9.5	**72**	34	-4.0	-2.0	**115**	54	-16.0	-0.0
30	14	-0.5	-9.0	**73**	34	-4.0	-2.0	**116**	55	-18.0	-0.0
31	14	-0.5	-9.0	**74**	35	-4.5	-2.0	**117**	55	-18.0	-0.0
32	15	-0.5	-8.5	**75**	35	-4.5	-2.0	**118**	56	-20.0	-0.0
33	15	-0.5	-8.5	**76**	36	-5.0	-2.0	**119**	56	-20.0	-0.0
34	16	-1.0	-8.0	**77**	36	-5.0	-2.0	**120**	57	-22.0	-0.0
35	16	-1.0	-8.0	**78**	37	-5.0	-1.5	**121**	57	-22.0	-0.0
36	17	-1.0	-7.5	**79**	37	-5.0	-1.5	**122**	58	-26.0	-0.0
37	17	-1.0	-7.5	**80**	38	-5.5	-1.5	**123**	58	-26.0	-0.0
38	18	-1.0	-7.0	**81**	38	-5.5	-1.5	**124**	59	-32.0	-0.0
39	18	-1.0	-7.0	**82**	39	-5.5	-1.5	**125**	59	-32.0	-0.0
40	19	-1.0	-6.5	**83**	39	-5.5	-1.5	**126**	60	OFF	-0.0
41	19	-1.0	-6.5	**84**	40	-6.0	-1.5	**127**	60	OFF	-0.0
42	20	-1.5	-6.0	**85**	40	-6.0	-1.5				

C.2 Hexadecimal Conversion Tables

Table 46. Hex to Decimal

	0	1	2	3	4	5	6	7	8	9	a	b	c	d	e	f
0:	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
10:	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
20:	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47
30:	48	49	50	51	52	53	54	55	56	57	58	59	60	61	62	63
40:	64	65	66	67	68	69	70	71	72	73	74	75	76	77	78	79
50:	80	81	82	83	84	85	86	87	88	89	90	91	92	93	94	95
60:	96	97	98	99	100	101	102	103	104	105	106	107	108	109	110	111
70:	112	113	114	115	116	117	118	119	120	121	122	123	124	125	126	127

Table 47. Decimal to Hex

	0	1	2	3	4	5	6	7	8	9
0:	0	1	2	3	4	5	6	7	8	9
10:	a	b	c	d	e	f	10	11	12	13
20:	14	15	16	17	18	19	1a	1b	1c	1d
30:	1e	1f	20	21	22	23	24	25	26	27
40:	28	29	2a	2b	2c	2d	2e	2f	30	31
50:	32	33	34	35	36	37	38	39	3a	3b
60:	3c	3d	3e	3f	40	41	42	43	44	45
70:	46	47	48	49	4a	4b	4c	4d	4e	4f
80:	50	51	52	53	54	55	56	57	58	59
90:	5a	5b	5c	5d	5e	5f	60	61	62	63
100:	64	65	66	67	68	69	6a	6b	6c	6d
110:	6e	6f	70	71	72	73	74	75	76	77
120:	78	79	7a	7b	7c	7d	7e	7f		

D. Glossary and Bibliography

Some terms used in this manual may not be familiar to you. Their definitions are presented in the following glossary. At the end of this chapter, you will find a short bibliography which is a good starting point for further research.

Many of the glossary items and their definitions are taken (with permission) from *The Audio Dictionary*, by Glenn White. In the interests of brevity, some of Glenn's definitions have been abridged and Glenn's extensive cross-referencing has been removed. Definitions taken from this book have the notation "(TAD)" appended to the definition. A more complete bibliographic entry for this book may be found in the bibliography.

D.1 Glossary

In this glossary, words typeset as follows: digital are cross-references to other words in this glossary. Glossary entries followed by "(TAD)" may be found in The Audio Dictionary.

Analog An audio signal is an electrical replica, or analog, of the waveform of the sound it represents. The voltage of the signal varies up and down (negatively and positively, in electrical terminology) the same way as the sound pressure varies up and down at the microphone.

As long as the signal is in this form, i.e., is a voltage that varies directly with the sound pressure, it is an analog, and audio devices which use such signals are analog devices. The majority of audio devices are analog in nature, though **digital** devices are increasing in popularity.

An analog audio device need not be electrical; the Edison mechanical phonograph was an analog device, the groove depth being an analog of the sound pressure at the recording diaphragm. (TAD)

AES/EBU A **digital** audio transmission system standardized by the Audio Engineering Society and the European Broadcast Union. An AES/EBU signal carries two audio channels as well as status information. The AES/EBU interface is balanced and uses XLR connectors. There are subtle differences in the actual signal format from the **S/PDIF** system.

AGC An automatic gain control (AGC) circuit adjusts the gain of an audio device in inverse proportion to the signal level entering the device. An example is a portable tape recorder which is designed for speech recording. When the talker is close to the microphone, the gain is reduced so as not to overload the tape. As the level from the talker decreases, for instance because of a greater distance, the **gain** increases to keep the recorded level the same.

This type of machine is often used for radio interviews, and usually the gain changes can be plainly heard as the background noise rises each time the speaker pauses for a few seconds, only to suddenly fall the moment the next syllable is uttered. (TAD)

A more recent meaning for AGC is the combination of the device described previously and a signal-sensing circuit that prevents the gain from changing when there is no valid signal present. This prevents the rising and falling background noise heard when a simple compressor is used as an AGC. The 602 uses this technique.

Analog to Digital Converter (ADC) In **digital** audio systems, the audio signal (**analog**) must first be converted to digital form before it can be further processed. This entails **sampling** the signal at very short successive time intervals, and converting the height of each sample to a digital word, which is simply a binary number indicating the amplitude of the waveform at that instant. *See also:* **quantization**)

The output of the A/D converter is a series of digital "words," expressed in binary form. Before the signal can be fed to an amplifier so it can be heard, it must undergo **digital-to-analog conversion**. This recovers a replica of the original audio signal from the digital words. (TAD)

Anti-Aliasing Filter Before a signal is subjected to the process of A/D conversion, it must be passed thorough a low-pass filter to remove any components that are higher in frequency than one-half the sampling frequency. This is because it requires at least two samples per cycle to determine the existence and strength of a frequency component, that is, it would require at least one hundred samples per second to encode a tone of 50 Hz. The A/D process will create spurious signal, called aliased components, if this rule is not followed.

In order to affect the audible signal as little as possible, an anti-aliasing filter is designed to be very steep, having an extremely rapid fall-off above the upper frequency limit. (TAD, abridged) *See also:* **brickwall filter, analog to digital converter, digital to analog converter**.

Anti-Imaging Filter In a digital audio system, in order to recover the signal from the digital words, a D/A converter is used. The output of this is a stair-step type of waveform which contains a great deal of high-frequency energy called "images." To reconstruct a smooth replica of the original signal, the stair-step is passed through a steep low-pass filter called an anti-imaging filter. It is similar, or even identical, to the anti-aliasing filter found at the input of the A/D converter, but its purpose is quite different. (TAD) *See also:* **brickwall filter, analog to digital converter, digital to analog converter**.

Attack Time Attack time is the time it takes for a **compressor** or **limiter** to reduce its gain after a strong signal is applied to it.

Transient signals which are shorter than the attack time of the device will not be affected by the gain reduction, so it is important that the attack time be as short as possible. (TAD) *See also:* **release time**.

Bandpass Filter A bandpass filter is a **filter** which has a **bandwidth**. Bandpass filters can be "broad," having a wide bandwidth, or "narrow," having a narrow bandwidth. They may be fixed in frequency and bandwidth, or variable in either frequency and/or bandwidth. (TAD)

Bandwidth The bandwidth of a **bandpass filter** is the upper cutoff frequency minus the lower cutoff frequency. It is thus the extent, in Hertz, of the frequency range, or band, passed by the filter.

Bandwidth is literally a frequency span, and is not necessarily connected to the specification of a filter. For instance, the human voice can be transmitted with good intelligibility if the **frequency response** of the transmission chain extends from about 100 Hz to about 3000 Hz. Thus, a 2900-Hz bandwidth is needed to transmit voice. This is about what a standard telephone system attains. The audio bandwidth, however is generally considered to be about 20 kilohertz. (TAD)

Brickwall Filter Some **lowpass filters** have such a steep cutoff slope that the graph of the slope resembles a brick wall (the slope of the sides, being vertical, is infinitely steep). Brickwall filters are commonly used for **anti-aliasing** and **anti-imaging**. (TAD)

Chorus An electronic music effect that modifies the sound of a single instrument to simulate a large group of the same instruments, for example, a vocal chorus or a string section. The subjective effect of a real chorus is caused by the fact that the many sound sources being mixed together all have slightly different frequencies and also do not have precisely steady frequencies. The mixture because extremely complex as the relative phases of the signals cause partial cancellation and reinforcement over a broad frequency spectrum.

The synthetic chorus effect was first attained by subjecting the input sound to a series of very short time delays and mixing the delayed sounds. The delays were then randomly varied, or modulated, to increase the uncertainty of the combined pitch. This could be called the "time domain" chorus synthesis and can be quite expensive if enough delay times are used to ensure a satisfactory result. A new type of chorus device operates in the frequency domain and is somewhat simpler and at the same time more convincing. The signal is split into many frequency bands by a series of **bandpass filters**, and each band is randomly varied in phase and amplitude, after which they are recombined. (TAD)

Clipping If a signal waveform is passed through an amplifier or other device which cannot accommodate its maximum voltage or current requirements, the waveform is sometimes said to be clipped, because it looks like it has had its peaks clipped by a pair of scissors. A clipped waveform contains a great deal of harmonic **distortion** and sounds very rough and harsh. Clipping is what typically happens when an audio amplifier output is overloaded or its input overdriven. The clipping point of an amplifier is defined as the maximum sine-wave signal level which, when viewed on an oscilloscope, shows no signs of flat-topping to the trained observer. (TAD)

Comb Filter A comb filter is a filter which has a series of very deep notches, or dips in its **frequency response**.; The spacing of the notches along the frequency axis is at multiples of the lowest frequency notch, so they look evenly spaced along a graph plotted on a linear frequency scale. On the more common logarithmic frequency scale, the notches become closer together on the paper as frequency increases.

A comb filter is produced when a signal is time-delayed and added to itself. Frequencies where the time delay is one-half the period and multiples of these frequencies are concealed when the signals are

combined because they have opposite polarity. If the signals are of equal strength, the cancellation is perfect and the notches are infinitely deep.(TAD) *See also:* **flanging**, **phasing**.

Compressor An audio device which reduces the **dynamic range** of a signal. The compressor is the first part of a compander (the combination of a compressor and **expander**).

The effect of the compressor is to make the loud parts of a signal softer and to make the very soft parts louder. Compressors are frequently used in recording popular music and in radio broadcasting, where very soft passages may be lost in the background noise of the listening environment. For instance, when music is playing on the radio in a car, the car's noise level will easily mask the quieter musical passages.

The **limiter** acts something like a compressor but operates only at the top end of the dynamic range. The subjective audibility of a compressor depends strongly on its time constants (attack and release times) and they are selected with care to minimize obvious "pumping" of the volume. To restore the original dynamics to a compressed signal, a volume expander can be used, but great care must be taken that the time constants, slopes, and thresholds match those of the compressor. (TAD)

Condenser (capacitor) microphone One of the earliest types of microphones to be invented after Dr. Lee DeForest invented the Audion amplifier in 1906 was the condenser microphone. Thomas Edison is sometimes credited with its invention, but this seems to be in doubt. At any rate, Wente, of Bell Telephone Labs, designed a condenser microphone in 1917 and introduced it commercially in 1931.

The condenser microphone is a very simple mechanical system, with almost no moving parts compared to other microphone types. It is simply a thin stretched diaphragm held very close to a metal disc called a backplate. This arrangement is an electrical capacitor, and it is given an electric charge by an external voltage source (polarizing voltage). When sound acts on the diaphragm, the pressure variations cause it to move slightly in response to the sound waveform. This causes the capacitance to vary in like manner, and because the charge is fixed, the voltage on the backplate will vary according to the laws governing the capacitor. This voltage variation is the signal output of the microphone. The condenser microphone has extremely high output impedance, and must be placed very near a preamplifier to avoid loss of the signal.

It is possible by special treatment of the backplate and by combining several microphone elements to attain various directional patterns, including bi-directional (figure 8), cardioid, and super-cardioid. (TAD) *See also:* **phantom power**.

dB, Decibel Literally, one tenth of a bel. The bel is named after Alexander Graham Bell (which is why the 'B' in dB is capitalized), and the number of bels is defined as the common logarithm of the ratio of two powers. Thus, two powers, one of which is ten times the other, will differ by 1 bel; 10 watts are 1 bel higher in level than 1 watt. A 360-horsepower car is 1 bel more powerful than a 36-horsepower motorcycle. Any power ratio may be expressed in bels, and it is important to note that only power ratios are allowed. a bel is a pure number with no dimensions.

Decibel reference quantities

Unit of Measurement	Reference	Remarks
dB	none	Only useful in a relative sense, i.e. "3 dB hotter."
dBm	1 mw, 600 ohms	1 mw, 600 ohms = 0.775V RMS
dBv	0.775V, open circuit	Note*:* **open circuit, small V**
dBV	1V, open circuit	Note*:* **capital V**
dBu	0.775V RMS, open circuit	Same as dBv. Becoming more common because of confusion between 'v' and 'V'.

The bel had its origin in the bell Telephone Labs, where workers needed a convenient way to express power losses in telephone lines as power ratios. Because the bel is a power ratio of 10, and this is a rather large ratio, it is convenient to divide it into tenths of bels, or decibels (abbr. dB). Ten dB is 1 bel; thus the decibel is ten times the common log of the ratio of two powers. The decibel was originally called the "transmission unit," or TU, by the Bell Labs people. (TAD, severely abridged)

The decibel is commonly used as a means of expressing audio signal levels. In dynamic-range processors, like compressors and limiters, their input to output relationship, or compression ratio, is a plot of the unit's input signal, in dB, to the unit's output signal, also in dB. Since the decibel represents a ratio of

two quantities, when discussing absolute signal levels, it is important to know what reference quantity was used:

De-Esser A de-esser is a special type of **compressor** that operates only at high frequencies, usually above 3 or 4 kHz. It is used, especially in the broadcast industry, to reduce the effect of vocal **sibilant** sounds, which are normally too strong when singers and announcers use very close-up microphones. When the high-frequency energy exceeds a preset **threshold,** the compressor starts to operate to reduce the high-frequency response. Low-level, high-frequency sounds are not reduced. (TAD)

The 602 uses a variation of this technique. The essential difference is that the threshold setting is relative to the ratio of sibilant to non-sibilant sounds. The compressor operates across the entire audio band; i.e. all signals are reduced in the presence of a sibilant sound that exceeds the preset threshold.

Digital The application of digital computer-based technology to the recording and reproduction of music is somewhat loosely called digital audio. (TAD)

Digital delay A digital device which provides an adjustable time delay. Time delays are used in artificial **reverberation** systems, for special echo effects in music recording, and to provide delayed sound to certain loudspeakers in some sound reinforcement systems. Before the advent of digital delays, the only the effect could be achieved was by tape **echo**, or by placing a loudspeaker at one of a long tube and a microphone at the other. This gives a delay of about 1 millisecond per foot of length, and it becomes bulky when long delays are needed.

Digital Signal Processing/DSP The manipulation and modification of signals in the digital domain (after having undergone analog-to-digital conversion). A great many electronic music instruments use DSP, as do certain test equipment types such as the FFT analyzer. Most DSP devices have a microprocessor inside them to do most of the work. (TAD)

The 602 uses two DSP 56001 digital signal processors and a 68HC11 microprocessor.

Digital to Analog Converter (DAC) The component within a **digital** audio device which converts binary digital words into an analog signal that can be amplified and sent to a loudspeaker, etc. The DAC is the last link in the digital chain, just before the anti-imaging filter. (TAD) *See also:* **analog to digital converter**.

Distortion Theoretically, any addition or modification to a signal caused by any type of equipment could be called "distortion," but the term has come to be somewhat more restricted in its use. Distortion may be conveniently grouped into six types:

1. Nonlinear distortion, manifested as harmonic distortion and intermodulation distortion. Harmonic distortion is the production of harmonics of the original signal by the equipment. Intermodulation distortion is the production of sum and difference products of the various frequency components that make up an audio signal.
2. Frequency distortion, the unequal amplification of different frequencies.
3. Phase distortion, an effect caused when phase shift in an audio device is not a linear function of frequency. In other words, different frequencies experience different time delays.
4. Transient distortion, including transient intermodulation distortion (TIM).
5. Scale distortion, or volume distortion.
6. Frequency modulation distortion. Examples of this are flutter and wow, and doppler distortion caused by the motion of loudspeaker cones. (TAD)

Downward Expander See Expander.

Dynamic Filter A dynamic filter is a type of single-pass noise reduction system that uses one or two **filters** whose cutoff frequencies are controlled by the level of the signal. As the signal level falls during soft passages, the high-frequency response is reduced (like turning down the treble tone control), and when the signal level is high, the full bandwidth is restored.

The effective operation of such a system depends on the fact that the noise will be masked by the signal during loud passages, and this is true in many, but by no means all, cases.

A key element in the design of dynamic filters is the choice of time constants during the time that the bandwidth is changing. If they are too fast, distortion results, and if too slow, the noise will be heard to swish in and out as the signal level changes. (TAD)

Dynamic Microphone A dynamic microphone consists of a diaphragm with a coil of wire attached to it such that sound pressure moving the diaphragm causes the coil to move in a magnetic field supplied by a permanent magnet. Motion of the coil causes an electric current to be induced in it, and this is the signal output of the microphone. It is similar to a dynamic loudspeaker operating in reverse. (TAD)

Dynamic Range The dynamic range of a sound is the ratio of the strongest, or loudest, part of the weakest, or softest, part; it is measured in **decibels**. A full orchestra may have a dynamic range of 90 dB, meaning the softest passages are 90 dB less powerful than the loudest ones. Dynamic range is a power ratio, and has nothing to do with the absolute level of the sound.

An audio signal also has a dynamic range, which is sometimes confused with signal-to-noise ratio. Rarely is the dynamic range of an audio system as large as the dynamic range of an orchestra because of several factors. The inherent noise of the recording medium determines the softest possible recorded sound, and the maximum signal capacity of the system (**clipping** level) limits the loudest possible sound. Many times an extremely wide dynamic range is not desirable (e.g., in radio broadcasting for listening in cars) and broadcasters frequently use **compressors** and **limiters** to reduce the dynamic range of the signals before they are transmitted. This type of signal processing distorts the music in a more or less noticeable way, symphonic music being most sensitive to it. (TAD)

Echo Commonly used incorrectly to mean **reverberation**, echo, technically is a discrete sound reflection arriving at least 50 milliseconds after the direct sound. It also must be significantly above the level of the reverberation at that time.

"Echo chambers" are reverberation rooms which are carefully designed to be without echoes. If an actual echo is desired in a recording, a tape recorder is sometimes used to add a time delay (tape delay), the delay representing the time it takes the tape to move between the record and reproduce heads. This is called "tape echo," and is appropriate usage. A popular way to get the same effect is to use a digital time delay system (**digital delay**), where the time delay is variable. (TAD)

Equalizer An equalizer, contrary to what its name implies, alters or distorts the relative strength of certain frequency ranges of an audio signal. In a sense, it should probably be called an "unequalizer." However, the first equalizers were used to make the energy at all frequencies equal, or to achieve "flat response," in telephone lines, and this is where the term originated. Another early use of equalizers was in the sound motion picture industry, where they were used to improve intelligibility in film sound tracks. Later on, equalizers were found useful for creating special sound effects in the early days of radio and movies, where they are extensively used to this day. All equalizers are made up of various circuits called **filters**, which are frequency-selective networks containing resistors (R), capacitors(C), and inductors (L). Normally filters attenuate certain frequency ranges and do not boost them; however some equalizers that boost the signal are called filters.

An equalizer can boost or attenuate a certain frequency band, but in common usage, equalize means to boost. The preferred terminology for the actual process is boost/cut rather than equalize/attenuate. In Britain preferred usage is lift/dip.

Equalizers that can have peaks in their response curves (such as **parametric** and graphic equalizers) are characterized by the relative sharpness of the peaks. The Q of a filter is a measure of this sharpness and is defined as the center frequency divided by the half-power bandwidth. For instance, a one-third octave filter centered at 1000 Hz will be 232 Hz wide at its half-power points. Its Q is thus 1000/232, or 4.31. Filters with Q values much higher than this tend to "ring," distorting transients, and call attention to themselves when used in sound systems. *See also:* **Q**, **parametric equalizer, shelving equalizer**. (TAD, abridged)

Expander A device for increasing the dynamic range and reducing the apparent noise of a signal. A volume expander decreases the system gain as the signal level decreases, making soft signals softer still. This results in an apparent noise decrease because the relative level between the softest and loudest sounds is greater. If the noise level is already low enough that the signal will mask it in the loud passages, the expansion will put the low end of the dynamic range at a point where the ear has reduced sensitivity, making the noise less audible (TAD). This definition for an expander is commonly used for downward expanders as well.

Filter A filter is a type of **equalizer** which is designed to reduce the energy at a certain frequency or in a certain frequency band. Filters always act as subtractive devices, never adding anything to a signal; at least they should not. The most common types of filters are **analog** filters, which operate on signals directly.

Digital filters operate on signals which have been digitized. They are purely mathematical, performing a series of arithmetic operations on the digital words. In a sense, digital filters are synthesized filters; digital techniques being used to emulate or simulate analog filters. Digital filters have the advantage of being drift-free. They always do their job in exactly the same way. They can be designed for nearly any desired characteristics in the frequency domain and in phase response. (TAD, abridged)

Flanging A special effect made popular in the 1960's where a delayed version of a signal is mixed with the signal, creating a "swooshing" sound.

Flanging was first done by recording a signal on two similar tape recorders, playing them back simultaneously, and mixing them together. The record-playback sequence on the tape recorders results in a small time delay of perhaps a tenth of a second. Both output signals are delayed by the same amount if the tape recorders are similar and they add together in the mixer, and the sound heard is essentially the same as the signal at the input to the tape recorders.

To achieve the effect of flanging, one recorder is slowed down a little, increasing its time delay. This is done by pressing one's thumb against the flange of the tape recorder supply reel, hence the name "flanging." When the time delay is different for the two combined signals, there will be frequencies where the phase shift is 180 degrees, and the signals will cancel, causing deep dips or holes in the frequency response curve. This is called the comb filter effect. As the speed is varied, the frequency of the dips is swept across the frequency range, giving the swooshing sound. Attaining the most desirable effect requires an educated thumb. The best effect is obtained when the signal being flanged contains frequencies over a wide range. (TAD) *See also:* **phasing**.

Frequency Response Also known as magnitude response, is the graph of the variation in output level of a device over frequency, with a constant amplitude input signal.

Full Scale When audio signals are converted to their digital equivalents using an **analog-to-digital converter,** the signal level that causes the output of the converter to reach its digital maximum is referred to as full scale.

Gate A circuit which performs like a switch, allowing a signal either to pass or not, is called a gate. The position of the gate (open or closed) is controlled by an applied voltage which can come from a number of different places. If the level of the signal determines the gate opening, it is a noise gate, closing when the signal level is so low that the noise would be audible. (TAD) A noise gate is an extreme example of a downward **expander**.

Group Delay The slope of the phase-versus-frequency curve of a frequency response function, that is, the rate of change of phase of the response as a function of frequency. Group delay is a property of a device or a system.

A pure time delay, equal at all frequencies, gives a constant slope of phase versus frequency. If in an audio component this slope is not constant, but varies with frequency, the component is said to produce group delay distortion. This is equivalent to a time delay that varies with frequency. For instance, an anti-aliasing filter will typically have a phase response curve which slopes sharply down at high frequencies, which means that the high-frequency components will be delayed longer in their passage through the filter. The audible result is a loss of precision in musical transients; they are spread out, or "smeared," in time, and a more diffuse stereo image results. (TAD)

Highpass Filter A highpass **filter** uniformly passes signals above a certain frequency, called the cutoff frequency. The cutoff frequency is where the filter response is 3 decibels below the nominal response. The response rolloff in the stopband may be gradual or sharp.

The "rumble" filter found in many record player systems is a highpass filter. (TAD) *See also:* **lowpass filter**.

Impedance In an electric circuit containing direct current, the magnitude of the current is determined by the voltage across the circuit divided by the resistance of the circuit. This is known as Ohm's law.

In a circuit containing alternating current, the situation is more complex; the "resistance" presented to the current is a function of frequency. This "AC resistance" is called impedance and is also measured in ohms. Impedance is the vector sum of resistance, capacitive reactance, and inductive reactance. Alternating currents are affected by resistance the same way as direct currents, and Ohm's law can be used for AC if the reactances are zero, that is, if there are no capacitors or inductors in the circuit. (TAD, abridged)

Limiter A special type of **compressor** which prevents the signal from exceeding a certain preset level (**threshold**), no matter what the input signal level may be. Limiters are sometimes used for special effects in popular recordings, especially vocals. A vocal with limiting will be essentially at the same level regardless of the effort put out by the singer, from a soft voice to a shout. The shouting will sound subjectively louder, however, because of the increased harmonic content of the sound. The **dynamic range** of a singer at a close range to a microphone is far greater than that of any instrument or musical ensemble, and when recording a vocal with an ensemble without limiting, a great deal of gain riding must be done to maintain musical balance.

Limiters are sometimes used in front of power amplifiers in sound reinforcement systems or radio transmitters to prevent unexpected high-level signals from causing overloading and large amounts of distortion. (TAD)

Lowpass Filter A filter which uniformly passes frequencies below a certain frequency called the cutoff frequency. Usually this is defined as the frequency where the amplitude response of the filter is 3 decibels below its nominal value.

Many early tone controls were variable lowpass filters. (TAD)

MIDI The Musical Instrument Digital Interface (MIDI) is a standard communications interface for use between electronic music synthesizers of various manufacture. (TAD)

MIDI is also used for controlling other peripheral devices, both musical and not. Besides synthesizers, audio signal processors (like the 602), electronic drums, and lighting controllers are now MIDI-controlled.

Noise Gate **See Gate.**

Overload An overload is said to occur when the input signal level in an audio device is so large that it drives the device out of its linear range and into **distortion** or **clipping**. Overload may be continuous or may occur only on short peaks in musical waveforms. The latter condition is common with certain waveforms, such as sharp percussive sounds which have a peak value much greater than their average value. This "peak clipping," as it is called, must be avoided for true high-fidelity recording and reproduction, although a small amount of it may be quite difficult to hear in practice. (TAD)

Oversampling In some **digital** audio components, the sampling frequency (44.1 kHz for compact discs) is raised to a multiple of that frequency. For example, if the sampling frequency were raised by a factor of 4, three artificial samples must be created in between each pair of original samples. These samples are zero in level, and they do not change the information content of the original samples. Digital filtering is then used for interpolation of the zero samples to values intermediate between the true sampled values. But because the sampling rate is now so much higher, a very gentle **anti-aliasing filter** can be used rather than the **brickwall filter** usually needed, resulting in much less phase distortion. *See also:* **sampling rate**. (TAD)

Pan (panpot) Short for panoramic potentiometer, which is two connected volume controls with a common knob so wired that as one is turned "up," the other is turned "down." If the stereo channels are controlled by a panpot, the apparent position of the sound will move from left to right as the control is turned. The balance control on most stereo amplifiers is actually a simple panpot.

Panpots are used in recording to place the apparent position of a sound, such as a soloist or other instrument, anywhere between the two loudspeakers. Its operation relies on the ability of our ears to localize a sound by level differences heard by our two ears. For a panpot to work properly, it must follow an accurately prescribed attenuation curve. In many recordings, several instruments are given separate positions by using a panpot on each one when the final "mix" is made. (TAD)

Parametric Equalizer An **equalizer** allowing control of center frequency, bandwidth, and boost/cut. *See also:* **shelving equalizer, peak/dip equalizer**.

Peak/Dip Equalizer An equalizer capable of providing a bandpass peak or dip (as differentiated from **shelving**) in its frequency response. Peak/dip equalizers are available in many forms, ranging from the program equalizers found on many mixing consoles, to graphic or **parametric** equalizers.

Phantom Power **Condenser microphone**s require a preamplifier to be close by due to the extremely high impedance of the microphone itself. This preamplifier is in the housing of the microphone, and it needs a power source. Sometimes a battery is used, but more often a multi-wire cable brings the audio signal from the microphone and brings the power from an external power supply to the preamp. This is a rather

bulky and expensive arrangement. To eliminate the multiconductor cable, frequently a scheme called phantom powering is used, whereby the preamp power is carried by the same two wires that carry the signal. The key to its operation is the fact that the signal is alternating current and the power is direct current, and they can be separated by the action of a transformer.

The voltage used for phantom powering is usually 48 volts, but it can vary from about 12 to 52 volts. Microphones which use the lower voltages have a regulator circuit to reduce the higher voltages so no harm is done when they are plugged into a 48-volt phantom power supply. There is a DIN standard (no. 45-596) which specifies in detail the requirements for phantom power. (TAD)

Phantom powering is a compatible system; suitably wired low-impedance microphones may be plugged directly into a phantom powered input without regard to the presence or absence of phantom power. The technique gets its name from the old telephone term, "Phantom Circuit," which was a method for creating a second circuit on an existing pair of telephone wires.

Phasing, phaser A phaser, or phase shifter, is a device which gives an effect similar to flanging, but with less depth. It works by shifting the phase of the signal and adding it back to the signal. This causes partial cancellation at frequencies where the phase shift approaches 180 degrees. Phasing is sometimes called skying in Britain. (TAD) *See also:* **flanging**.

P-Pop A p-pop (“pee-pop) is the burst of air caused by uttering the letter “P”. Spoken directly into a **single-d** microphone, this blast of air usually causes a loud, audible popping sound. A windscreen of some sort is a good cure for p-popping as is an omnidirectional microphone. A skilled announcer will soften the initial attack of the letter, or turn their head slightly so as to avoid the microphone.

Proximity Effect Proximity effect is the increase in low-frequency sensitivity of a microphone when the sound source is close to the microphone. It is a characteristic of directional microphones, and some are much worse than others.

Proximity effect is a shortcoming, but sometimes it can be used to advantage. If a directional microphone is placed close to a bass instrument, the low tones will be enhanced, which could be advantageous for some music. A singer placed close to a directional microphone will sound much "bassier," and improvement in some voices, I suppose. Some of the early radio "crooners" and radio announcers used proximity effect to deepen and enrich their voices, and many frequently still do. (TAD)

Q In reference to a resonant mechanical or electrical circuit or a capacitor, Q stands for "quality factor." In the case of a resonant system, Q is a measure of the sharpness of the resonant peak in the frequency response of the system and is inversely proportional to the damping in the system. Equalizers that contain resonant circuits are rated by their Q value*:* the higher the Q, the higher and more well-defined the peak in the response. (TAD)

Quantization The representation of a continuous voltage span by a number of discrete values. Quantization is inherent in any digital audio system, and it adds quantization error, noise, and distortion to the signal.

The signal after quantization has a "staircase" shape rather than a continuous curve, and the difference between this and the original signal is quantization error. The amount of error will always be within one least-significant-bit (LSB); therefore the smaller the LSB, the better. In quantization of a sine wave, whose frequency is a submultiple of the sampling frequency, the error will have a definite pattern which repeats at a frequency of the signal. Thus, it will have a frequency content consisting of multiples of this frequency, and it can be considered as harmonic distortion rather than noise.

For music, however, the signal is constantly changing, and no such regularity exists. The quantization error is then wideband noise, and is called quantization noise. Quantization noise is difficult to measure because it does not exist without a signal. A sine test signal is not good because sometimes this results in distortion, not noise. If the sinewave frequency is chosen so it is not a submultiple of the sampling frequency, the quantization errors will be more nearly randomized and will resemble random noise. (TAD)

Ratio Short for compression ratio or expansion ratio. The term stands for the ratio of the change in the input signal of a device to the change in the devices output. When graphed on linear-scaled graph paper, the result is the familiar compression ratio curve (assuming the device is a compressor).

Although the term is most commonly used for compressors and expanders, there is no reason why it cannot be used for any device that alters its gain in some signal-level dependent manner (i.e. de-essers, limiters, noise-reducers, etc.).

Release Time The release time of a dynamics processor is the time required for the processor's gain to return to its nominal value, after the controlled signal exceeds (or doesn't exceed) a preset threshold. *See also:* **attack time, compressor, expander**.

Reverberation The remainder of sound that exists in a room after the source of sound is stopped is called reverberation, sometimes mistakenly called "**echo**." The time of reverberation is defined as the time it takes for the sound pressure level to decay to one-millionth of its former value. This is a 60-decibel reduction in level.

All rooms have some reverberation, and an important subjective quality of a room is its reverberation time, although other factors, such as ratio of direct to reverberant sound, are probably more important. In a real room, the sound heard by a listener is a mixture of direct sound from the source and reverberant sound from the room. Reverberant sound is diffuse, coming from random directions, and the direct sound allows us to localize the source of they sound. (TAD, abridged)

Ribbon Microphone, Velocity Microphone A type of microphone which usually has a polar pattern shaped like a figure 8. The first velocity microphone was the ribbon microphone, invented about 1931 by Harry F. Olson of RCA research laboratories.

The ribbon microphone uses as an active element a small corrugated strip of very thin aluminum ribbon hanging loosely in a strong magnetic field. The ribbon is moved by the action of air molecules, which are set in motion by the sound wave. the resonant frequency of the ribbon is very low, below the audible range, so the motion of the ribbon is "mass controlled," or is proportional to the velocity of the air particles. For this reason, it is called a "velocity microphone". (TAD) The motion of the ribbon within the magnetic field generates electricity, which is the microphone's output signal.

S/PDIF An acronym standing for Sony-Phillips Digital Interface Format. This term describes an interconnection standard/method commonly used for consumer-grade digital audio devices. The reason Sony and Phillips are jointly named is because they are the two companies that developed the Compact Disc. S/PDIF signals carry two audio channels as well as status information. The signal is unbalanced and RCA connectors are typically used for interconnection between devices. The format of S/PDIF signals is somewhat similar to the AES/EBU format.

Sampling Rate In a digital audio system, the audio signal must be fed into an **analog-to-digital converter** (ADC) to be changed into a series of numbers for further processing by the system. The first step in this is sampling, where the instantaneous signal amplitude is determined at very short intervals of time. **Sampling** must be done very accurately to avoid adding distortion to the digitized signal. The sampling rate, which is the number of samples per second, must be uniform and precisely controlled. (TAD, abridged) see also*:* **quantization**.

Shelving Equalizer An **equalizer** whose **frequency response** curves rise (or fall) to a maximum value, remaining at that value to the limits of audibility. The bass and treble controls on most home stereo amplifiers are shelving equalizers.

Sibilance Vocal recordings, especially if made with very close microphones, are often characterized by excessive loudness of the voice sibilants, and this effect is sometimes called "sibilance." The most difficult sibilants to reproduce accurately are the sounds "s" and "sh." (TAD) see also*:* **de-esser**.

Single-D Microphone A single-D microphone is a directional microphone having only one entrance for off-axis sounds. Single-D microphones exhibit a property called **proximity effect**, which is a boosting of low frequencies when the microphone is used close-up to the sound source. *See also:* **variable-D microphone**.

Slapback, Slap Echo The single repetition of a signal at a fixed time delay to simulate an echo from a single reflecting surface, as opposed to a multiple echo from a time delay, where the delayed signal is repeatedly fed back into the delay input. (TAD)

Sysex A **MIDI** message (command) that stands for System Exclusive. MIDI sysex messages are commonly used for controlling audio processors or other MIDI instruments. The sysex message exists to allow programming/controlling beyond that which is predefined in the MIDI specification.

Threshold A parameter commonly associated with dynamics processor and used to refer to a signal level at which processing begins or ends. In a compressor, the threshold level is that signal level where the change in level at the output no longer equals the change in level at the input.

Variable-D Microphone A variable-D microphone is a directional microphone having a multiplicity of entrances for off-axis sounds. Variable-D microphones exhibit proximity effect, although not to the degree that **single-D microphones** do. The term *Variable-D* is a trademark of Electro-Voice.

D.2 Bibliography

For further research the following books may be useful.

The Audio Dictionary, Second edition. Glenn D. White, Copyright 1991, University of Washington Press, Seattle Washington. This revised edition contains extended definitions of many of the terms used in this manual. In addition, the appendices should provide many hours of enjoyable reading. The book is available directly from the publisher, (206) 543 8870,, or from Old Colony Sound. (603) 924-6371.

Principles of Digital Audio. Ken C. Pohlmann, Copyright 1989, Howard W. Sams & Company, Indianapolis, IN. This is a good reference on the ins and outs of digital audio. The author covers basics as well as advanced topics. The book is available at major booksellers (like Tower Books), or from the publisher. (317) 298-5699.

The following publications are available from the Audio Engineering Society, 60 E 42nd Street, New York, NY 10165-2520. (212) 661-8528.

Digital Audio: Collected Papers from the AES Premiere Conference, Rye New York, 1982.

Present and Future of Digital Audio, Tokyo Japan, 1985.

Music and Digital Technology, Los Angeles, CA, 1987.

The Journal of the Audio Engineering Society, published monthly.

Notes

E. Architect's and Engineer's Specification

The integrated signal processor (ISP) shall be a dual input, dual output model accepting line-level signals, applying frequency response equalization, delay-based effects, and signal dynamics processing to that signal, and delivering the processed input signal to two outputs. All signal processing (equalization, delay, dynamics) shall take place in the digital domain. The ISP shall occupy one rack space (1U).

The equalizer block shall take the form of a user and MIDI programmable parametric equalizer capable of operating at three inflection points simultaneously. Band 1 of the equalizer shall be switchable between a lowpass shelving characteristic or a peak/dip characteristic. Band 3 of the equalizer shall be switchable between a highpass shelving characteristic or a peak/dip characteristic. All three bands of the equalizer shall be capable of operating over the following frequency ranges and bandwidths:

> 31 to 21.11 kHz with a bandwidth of .05 to 3 octaves, with a boost/cut range of +15 dB to -50 dB.

The delay block shall provide two delays capable of up to 330 milliseconds of delay. The delays shall be user and MIDI programmable. The feedback path for delay recirculation shall be cross-coupled between the two delays and the delay time shall be capable of accepting modulation either from an internal random number generator or from an internal sine- or triangle-wave source. The delay time shall be independently adjustable for each delay and provision shall be made to allow adjusting the delay times simultaneously while maintaining an offset in the delay times.

The dynamics block shall provide the following functionality: De-ess, Dynamic noise filter, Compressor, AGC/Leveler and Downward Expander. Within the dynamics block all sections are user and MIDI programmable and each dynamics function shall provide the following features:

De-Ess	High-ratio compression driven by a high-frequency selective sidechain. The de-esser shall provide a threshold control for user adjustment.
Dynamic Noise Filter	Sliding high-frequency rolloff controlled by the HF energy content of the input signal. The DNF shall provide threshold and frequency controls.
Compressor	Compression up to 10:1 ratio. The compressor shall provide threshold, ratio, attack, and release controls. The compressor characteristic shall be changeable between a hard-knee curve and a soft-knee curve.
AGC/Leveler	AGC over a 70 dB range, with adjustable gain platform and up to 4:1 ratio. AGC shall provide auto-release threshold, ratio, attack, and release controls.
Downward Expander	Downward expansion with up to 1:8 ratio. Expander shall provide threshold, ratio, attack and release controls.

The output block shall provide level and panning for the output signal. Both functions are user or MIDI programmable. The level control shall operate in the digital domain over a +/- 18 dB range. The panpot shall also operate in the digital domain with a sine-cosine characteristic law.

The ISP shall provide easy access to all user functions via a non-hierarchical parameter selection and modification scheme. There shall be a minimum of menus. Every major parameter shall be accessible via a button-press and subsequent adjustment of the parameter wheel.

The ISP shall provide a full MIDI implementation with the unit responding to the following messages:

- Program Change
- Control Change
- Pitch Bend
- After Touch
- System Exclusive (Sysex)

The MIDI implementation, via MIDI Sysex, Control Change, and Program Change, shall provide access to all major operating parameters of the ISP and real-time editing capabilities shall be provided to allow real-time parameter change during operation..

The ISP shall be capable of accepting line-level signals ranging from -4 to +18 dBu. The line input characteristics shall be 20 kΩ balanced bridging.

The ISP shall be capable of accepting and delivering digital input signals at either a 44.1 kHz or 48.0 kHz sample rate. The ISP shall be capable of converting analog signals to digital form using either the 44.1 kHz or 48.0 kHz sample rates.

The ISP shall be capable of accepting digital input signals conforming to the AES/EBU standard or to the S/PDIF standard. Two such digital inputs shall be provided. The digital inputs shall utilize a 3-pin XLR female connector and an RCA connector. The digital inputs shall conform to the AES/EBU standard and S/PDIF standard respectively.

The ISP shall be capable of delivering digital output signals conforming to the AES/EBU or S/PDIF standard. Two such digital outputs shall be provided. The digital outputs shall utilize a 3-pin XLR male connector or an RCA connector. The digital outputs shall conform to the AES/EBU standard and S/PDIF standard respectively.

The analog inputs shall be active balanced bridging designs. The line inputs shall be terminated in 3-pin XLR female connectors. All analog input circuitry shall incorporate RFI filters. The analog outputs shall be active balanced designs having equal source impedances and terminated with 3-pin XLR male connectors. All XLR connectors used for analog input/output shall conform to the AES/IEC polarity standard.

The balanced inputs shall accommodate +22 dBu signals without distortion, and the balanced outputs shall be capable of delivering +21.5 dBm into a 600 ohm load.

The ISP shall be capable of operating by means of its own built-in power supply connected to 117V nominal ac (105-130V) 50/60 Hz, 20 watts (230V nominal, 207-253V ac, 50 Hz where applicable).

The unit shall be a Symetrix Incorporated model 602 Stereo Digital Processor

F. Disassembly Instructions

Caution: *These servicing instructions are for use by qualified personnel only. To avoid electric shock do not perform any servicing other than that contained in the operating instructions portion of this manual unless you are qualified to do so. Refer all servicing to qualified service personnel.*

Caution: *Parts of the 602 use surface-mounted semiconductors. Removing and replacing these parts requires special tooling and special techniques. This is not a job for the faint-of-heart nor is it something that you should attempt for the first, second (or even third) time Do not attempt this at home!. We strongly advise that you should refer all servicing to the factory.* ***Any damage to the 602 that, in our sole opinion, resulted from improper surface-mount technique or improper tooling is not covered by the warranty.***

Warning: Lethal voltages are present inside the chassis. Perform all service work with the unit disconnected from all AC power.

F.1 Top Cover Removal

1. Ensure that the 602 is disconnected from the AC power source.
2. Remove three 6-32 x 1/2 inch screws from each side of the chassis.
3. Remove two 6-32 x X 1/4 inch flag-head screws from the top cover
4. Remove four 6-32 x 1/2 button-head screws (you'll need a 5/64 inch allen wrench).
5. Remove one 6-32 x 1/2 inch button-head screw from the top-middle of the front panel.
6. Lift the top cover free of the chassis.

F.2 Circuit Board Removal

There are five circuit boards inside the 602.

> **Caution**: *The circuitry within the 602 is static sensitive. Use appropriate techniques to eliminate static electricity from your body and from the surrounding area. If these techniques are not familiar to you, you should refer servicing of your 602 to the factory.*

1. Ensure that the 602 is disconnected from the AC power source.
2. Remove the top cover using the procedure described previously.

F.2.1 Analog Board Removal

1. Rotate the two gain controls until you can see the two setscrews on the shaft coupler near the circuit board mounted potentiometer. Loosen the screw located towards the front of the unit and slide the knob and shaft out of the chassis. Repeat for the remaining gain control.
2. Remove the two 6-32 x 1/4 screws from the front edge of the circuit board.
3. Disconnect the ribbon-wire jumper between the analog and digital boards. It is sufficient to remove/disconnect only one end.
4. Remove four 6-32 x 1/4 screws from the shield surrounding the analog board. Remove the shield.
5. Unlock the inserts within the four XLR connectors. (see procedure elsewhere in this section).
6. Slide the analog board towards the front of the unit, then lift it clear of the chassis after the connector bodies clear the connector shells. It may help to push on the connector bodies from the rear of the chassis.

F.2.2 Digital Board Removal

Caution

The circuitry within the 602 is static sensitive. Use appropriate techniques to eliminate static electricity from your body and from the surrounding area. If these techniques are not familiar to you, you should refer servicing of your 602 to the factory.

1. Disconnect the ribbon-wire jumper between the analog and digital boards. It is sufficient to remove/disconnect only one end. Disconnect the ribbon wire connectors connected to the Wheel, and to the front-panel circuit board. Disconnect the power supply connector located at the front-right of the unit.
2. Remove three 6-32 x 1/4 inch screws from the digital circuit board.
3. Unlock the XLR connector inserts using the procedure found elsewhere in this section.
4. Slide the digital board towards the front of the unit, then lift it clear of the chassis after the connector bodies clear the connector shells. It may help to push on the connector bodies from the rear of the chassis.

F.2.3 Power Supply Board Removal

1. Disconnect the power supply connector located at the front-right of the unit.
2. Remove four 6-32 x 1/4 inch screws from the power supply circuit board.
3. Remove two 6-32 x 1/4 inch screws from the heatsink attached to U5.
4. Remove two 6-32 x 1/2 inch screws from the IEC power connector at the rear panel.
5. Disconnect the green chassis ground wire, at the chassis, by removing the nut securing it to the chassis stud.
6. Slide the power supply board towards the front of the unit, then lift it clear of the chassis.

F.2.4 Front Panel Board Removal

1. Disconnect all of the ribbon wire connectors from the digital board.
2. Loosen the setscrew on the Wheel and remove it.
3. Rotate the two gain controls until you can see the two setscrews on the shaft coupler near the circuit board mounted potentiometer. Loosen the screw located towards the front of the unit and slide the knob and shaft out of the chassis. Repeat for the remaining gain control.
4. Remove the five button-head screws securing the front panel. Remove the front panel.
5. Remove the five 6-32 x 1/4 inch screws securing the sub-front panel to the chassis.
6. Lift the front panel board clear of the chassis.

F.3 XLR Connector Removal (Important!!)

The XLR connectors must be disassembled prior to removing either the analog or digital boards from the chassis. After disassembly, the connector body remains riveted to the chassis and the connector body remains with the circuit board.

Unlock the XLR connectors by inserting a 2mm slot-head screwdriver into the hole located between pins 1 and 2 of the XLR connector insert. Twist the retaining lug CCW to unlock the connector body. When separating the connector body from the shell, it may help to push on the connector body from the outside (through the shell).

G. Presets and Other Stuff

This appendix contains material that defies categorization or inclusion elsewhere. You'll find things like the Preset Programs list, a Programmer's Worksheet, and the MIDI implementation table.

Symetrix 602 Programmer's Worksheet

Program Number	Programmer:		Comments: Table revised 4/4/94
Program Name:			
Source: ❒ Dig. ❒ CH1 ❒ CH2 ❒ Stereo			
CH1 Gain	CH2 Gain		

FILTER	Freq	Level	Bandwidth	Freq/BW ROC	Level ROC
Filter Block 1 ❒ Pk ❒ Shlf	Hz	dB	Oct.		
Filter Block 2 Peak only	Hz	dB	Oct.		
Filter Block 3 ❒ Pk ❒ Shlf	Hz	dB	Oct.		

DE-ESS	Attk.	Rel.	Abs. Thr.	Rel. Thr.	NR	Rel. Thr.	Abs. Thr.	Freq.
❒ IN ❒ OUT	ms	ms	dB	dB	❒ IN ❒ OUT	dB	dB	Hz

DYNAMICS	Status	Attack	Release	Ratio	Threshold
Expander	❒ IN ❒ OUT	ms	ms		dB
Compressor	❒ IN ❒ OUT	ms	ms		dB
AGC	❒ IN ❒ OUT	ms	ms		dB
SEt Param.					
SEt Param.					

DELAY	Mix	Delay 1	Delay 2	Feedback	Filter
Mod Rate	Mod Depth	Level ROC	Delay ROC	Osc Type	
				❒ Random ❒ Sine ❒ Triangle	

OUTPUT	Output Level	Output Pan	Output ROC
Output			

REALTIME	Source	Scale Factor	Offset	Parameter #
Block 1				
Block 2				

Note: ROC = Rate Of Change

Rev 1.1, 11/15/94

Model: Symetrix 602
Device Type: 02

MIDI Implementation Chart

Doc: Rev 1.1, 11/15/94

Midi Manufacturer ID: 00, 00, 5E

Functio= n		**Transmitted**	**Recognized**	**Remarks**
Basic Channel	Default Channel	1-16 1-16	1-16 1-16	Memorized
Mode	Default Messages Altered	**X** **X** **X**	**X** **X** **X**	Memorized OMNI ON/OFF
Note Number	True Voice	**X** **X**	**X** **X**	
Velocity	Note ON Note OFF	**X** **X**	**X** **X**	
After Touch	Key's Ch's	**X**	**O**	Realtime MIDI blocks
Pitch Bender		**X**	**O**	Realtime MIDI blocks
Control Change		**X**	07 Volume 10 Pan 32 Bank Select	Any using realtime MIDI blocks
Program Change	True#	**X**	**O** 0-127	Program# 1-128
System Exclusive		**O**	**O**	
System Common	:Song Pos :Song Sel :Tune	**X** **X** **X**	**X** **X** **X**	
System Real Time	:Clock :Commands	**X** **X**	**X** **X**	
Aux Messages	:Local ON/OFF :All Notes OFF :Active Sense :Reset	**X** **X** **X**	**X** **X** **X**	
Notes				

Mode 1: OMNI ON, POLY
Mode 2: OMNI ON, MONO
Mode 3: OMNI OFF, POLY
Mode 4: OMNI OFF, MONO

O : Yes
X: No

G.3 Presets and Building Blocks

The following table lists every factory-supplied program in the 602. Note that those programs listed in the RAM column may be modified or overwritten by another (possibly totally different) program and those programs listed in the ROM programs may be modified, but not saved, except in unprotected RAM memory.

Programs 100 through 128 are building-block programs. These programs give you quick setups for certain parameters that involve realtime MIDI or those parameters that are not directly accessible from the front panel. Use the programs to start your own.

Voice - Speech and Song			
RAM	**ROM**	**Name**	**Description**
1	129	TLM-170 Male	Neumann TLM-170 optimized for male voice
2	130	Maximum Intelligibility	EQ set for voice range boost
3	131	Talk Show Announcer	Small room ambience
4	132	TLM-170 Speech+Delay	Neumann microphone with echoes (short)
5	133	TLM-170 Speech+Long Delay	TLM-170 microphone with echoes (long)
6	134	Speech+Chorus	EQ optimized for speech and slight chorus effect
7	135	Clear Vox	Small room ambience with bright EQ settings
8	136	Metal Closet	Simulates small voice over booth
9	137	Speech Leveler	AGC with low ARM sensitivity
10	138	AGC-Leveler	Program to simulate the Symetrix 421 AGC-Leveler
11	139	TLM-170 Female	Neumann TLM-170 optimized for female speech
12	140	Vocal Flange Female	Female EQ settings with light flanger effect
13	141	Supremes	Bright EQ settings with flutter echoes
14	142	Speech Leveling Female	AGC program with EQ set for female speech
15	143	Aggressive, bright narration	AGC set for intelligibility
16	144	Mellow narration	Close, intimate sound
17	145	FM Disc Jockey 1	Morning show
18	146	FM Disc Jockey 2	Drive time show
19	147	FM Disc Jockey 3	Midnight program
20	148	Wide stereo image voice	Voice image widened
21	149	Vocal - Handheld Mic	EV BK-1 Microphone with EQ set for live performance
22	150	Vocal - Handheld Mic with Echo	EV BK-1 Microphone with delay
23	151	Vocal Flange	EQ set for Voice with Flange
24	152	Elvis	EQ set for the King with simulation of tape delay
25	153	Pop Vocal	Set for live style performance with delay
26	154	TV Commercial Announcer	Institutional voice
27	155	Stereo TV Commercial Announcer	Widened image
28	156	50's Country Vocal	A touch of twang
29	157	Rock/blues Vocals	A bit of echo
30	158	Handheld Vox Female	AKG C-535EB microphone with EQ set for Female voice
31	159	Handheld Vox Female/Delay	AKG C-535EB Microphone with delay settings
32	160	Background Vocals	Slight delay for width
33	161	Thin, Airy Vocals	Harsh EQ for effect
34	162	High-definition Vocals	EQ set for vocal strength
35	163	Rock Echo Vocals	EQ set to cut through mix
36	164	Doubled Rock Vocals	Delay set for doubling
37	165	Tripled Rock Vocals	Stereo delay set at different intervals
38	166	Voice Thickener/Warmer 1	Adds stereo width
39	167	Voice Thickener/Warmer 2	More stereo width

Instruments - Mic and Line Inputs			
RAM	**ROM**	**Name**	**Description**
40	168	Flute W/Chorus	EQ set for flute with light chorus effect
41	169	Pound Guitar	signals above the threshold of the Dynamics section will modulate the chorus effect
42	170	Snare	Fat, bright sound for metal snare
43	171	Kick	Deep, round sound for kick drum
44	172	Electric Piano	Use line input / adds chorus effect
45	173	Acoustic guitar	Bright setting for steel strings
46	174	piano	EQ set for Grand Piano
47	175	Brass	Compressor set for containment/ slight delay for ambience
48	176	E Guitar Chet	Acoustic guitar program
49	177	Chorus Piano	Chorus effect for acoustic piano
50	178	Bass w/chorus	Use line input / EQ set for Bass w/chorus effect
51	179	FX guitar	Long echoes for electric guitar
52	180	Acoustic GTR w/chorus wash	tight chorus effect for guitar
53	181	Electric Bass/slap	Compressor set for bass slaps w/chorus effect
54	182	Solo Acoustic Guitar	Guild G37 W/Audio Technica 4033 microphone
55	183	Rhythm Acoustic Guitar	Warm EQ settings
56	184	Fingered Guitar	EQ set for plucked guitar
57	185	12 String Guitar w/Chorus	Wide Chorus for 12 String Guitar
58	186	E Guitar Chorus 1	Fat chorus for electric guitar
59	187	E Guitar Chorus 2	Really fat chorus for E guitar
Sweetening and Effects			
RAM	**ROM**	**Name**	**Description**
60	188	Bad Audio Restoration	EQ set to brighten signal/stereo image widening
61	189	Tape Transfer - Mastering	Light compression
62	190	Tape Transfer - Voice	De-essing and noise reduction/EQ set for voice
63	191	Basic Echo	Adjust Feedback control for more or less echoes
64	192	Random Pitch Echoes	Moving echoes
65	193	Big Troll Room	Makes voices sound large
66	194	Troll	Signal changes pitch and echoes
67	195	Leslie Simulator	Level and Frequency are modulated to the delay oscillator, adjust rate for faster or slower rotation
68	196	Auto-Pan Slow	Output level modulated by the delay section oscillator
69	197	Slow Effects Fade	ROC of Filters set to long fade once program is loaded
70	198	Falling Pitch Trail	Echoes change pitch
71	199	Unison Singalong	Echoes create "unison" voice that follows signal
72	200	Telephone Simulator	Thin sound for telephone like response
73	201	Computer Voice	Resonant robot-like voice
74	202	Darth Vader	Big, scary sound
75	203	Munchkins	Splattered echoes
76	204	64 Funny Cars	Repeated echoes
77	205	Stereo Randomizer	Creates moving stereo image
78	206	Quasi-Stereo	Simulates Stereo signal from mono source
79	207	Seventies Flanger	Deep, resonant Flanger

RAM	ROM	Name	Description
80	208	Small Semi-live Room	Simulated room ambience
81	209	Presentation Room	Medium size room
82	210	Gym P.A. System	Large size room
83	211	Backstage Interview	Medium size room with signal toward the front
84	212	Soft&Dry, Loud&Wet	Loud signals cause longer echoes/Delay Mix attached to Dynamics section threshold
85	213	Acoustic Chorus	Light Chorus effect
86	214	Public Address	Big PA simulation
87	215	Airport PA	Great for paging
88	216	Mondo Bizarro	Mixed up delays
89	217	Voice of Doom	Big voice sound
90	218	Be Bop A Lula	Swing style voice sound
91	219	Stereo Robots	Voice resonance
92	220	Stereo Invaders	Modulated effects
93	221	Telephone voice	Simulates radio call in show
94	222	Delay Chorus 1	Delay 1: 44ms, Delay 2: 62ms
95	223	Delay Chorus 2	Delay 1: 86ms, Delay 2: 104ms
96	224	Perspective 1	Front-center, slow cross fade
97	225	Perspective 2	Mid-right, slow cross fade
98	226	Perspective 3	Far-center, slow cross fade
99	227	Perspective 4	Mid-left, slow cross fade
Building Blocks - Used to create programs with specialized functions			
RAM	**ROM**	**Name**	**Description**
100	228	Panner	Output Levels attached to oscillator/adjust rate control for faster or slower panning
101	229	Doppler effect	Output level and band 1 frequency attached to oscillator
102	230	Compressor Sidechain	Filter 30Hz shelf
103	231	Compressor Sidechain	200Hz shelf
104	232	Compressor Sidechain	500Hz shelf
105	233	Compressor Signal Delay	Look ahead compressor with signal delayed - Maximum
106	234	Compressor Signal Delay	look ahead compressor with signal delayed Minimum
107	235	AGC ARM Sensitivity	Low 1.0
108	236	AGC ARM Sensitivity	Medium .5
109	237	AGC ARM Sensitivity	High .3
110	238	De-esser Absolute Threshold	Medium -50
111	239	De-esser Absolute Threshold	Low -60
112	240	AGC Signal Threshold	High -54
113	241	AGC Signal Threshold	Medium -66
114	242	AGC Signal Threshold	Low -74
115	243	Global Time Constants	Long/ P#8,P#14,P#20,P#70, 120
116	244	Global Time Constants	Long/ P#8,P#14,P#20,P#70, 96
117	245	Global Time Constants	Medium/ P#8,P#14,P#20,P#70, 58
118	246	Global Time Constants	Fast/ P#8,P#14,P#20,P#70, 31
119	247	Dynamics Preferences	Hard Knee Compressor/Hard Knee Expander
120	248	Dynamics Preferences	Medium Knee Compressor/Hard Knee Expander
121	249	Dynamics Preferences	Soft Knee Compressor/Hard Knee Expander
122	250	Dynamics Preferences	Hard Knee Compressor/Medium Knee Expander

RAM	ROM	Name	Description
122	250	Dynamics Preferences	Hard Knee Compressor/Medium Knee Expander
123	251	Dynamics Preferences	Medium Knee Compressor/Medium Knee Expander
124	252	Dynamics Preferences	Soft Knee Compressor/Medium Knee Expander
125	253	Dynamics Preferences	Hard Knee Compressor/Soft Knee Expander
126	254	Dynamics Preferences	Medium Knee Compressor/Soft Knee Expander
127	255	Dynamics Preferences	Soft Knee Compressor/Soft Knee Expander
128	256	Initialization Program	Used to "zero" any user program (1-127) in the 601 for creating new programs or for starting over.

Note: Program 256 works by overwriting the selected RAM location (1-127) with a set of rational settings. Use this program to create a fresh starting point for a program of your own or for when one of your programming efforts turns into Godzilla and must be destroyed.

Some of the preset programs modify parameters (like the compressor knee) that are accessed via the real time editor. For this reason, a program built on one of the factory programs (other than program 256) may work differently than one built on program 256. It is a good idea to find a preset program that is close to what you want, then modify it and save it.

Notes